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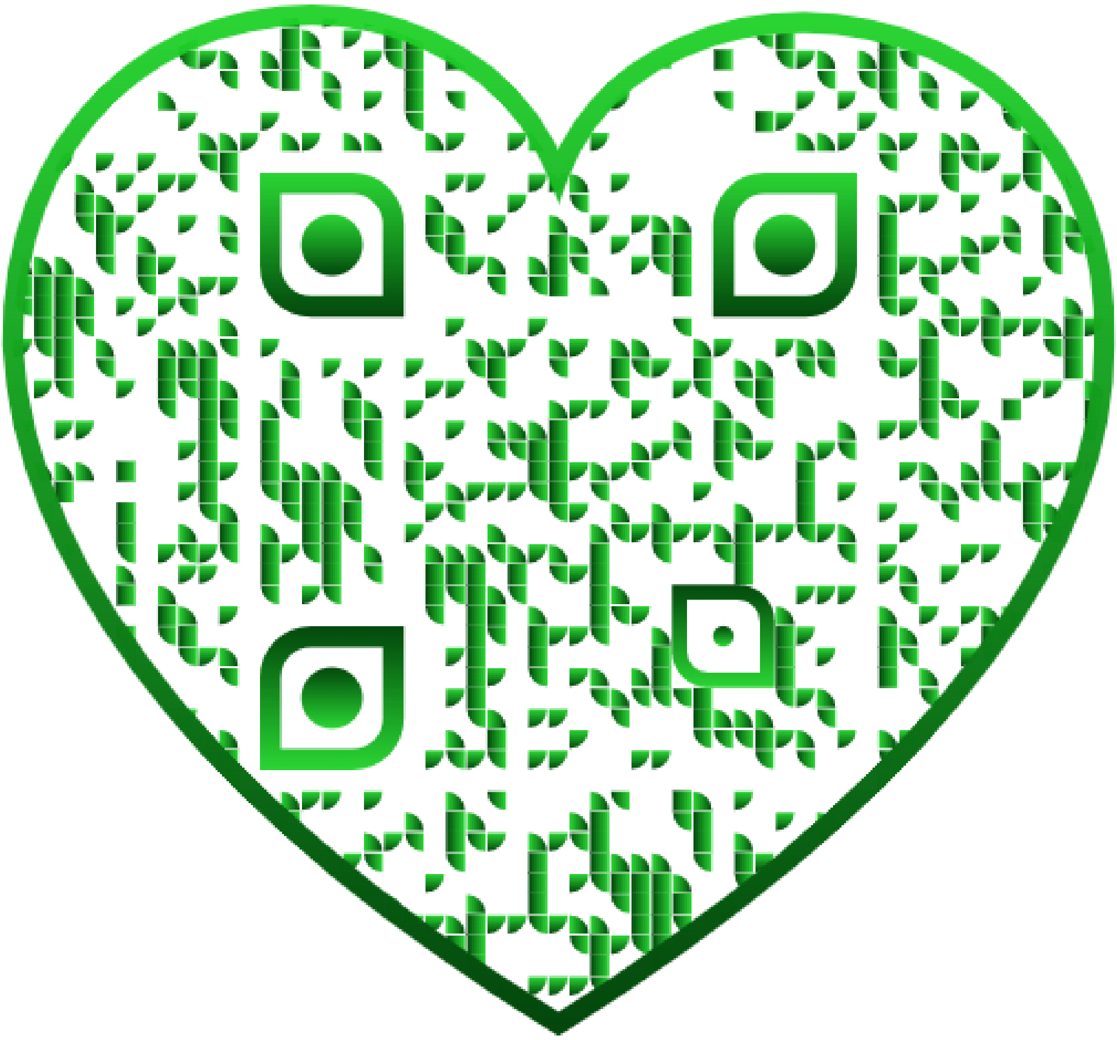

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Head Office:
KCK Media Corp.
P.O. Box 417
Chase City, VA 23924, US
Phone: 434-533-0246
Fax: 888-980-1303

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Strategic Media Marketing, Inc.
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Gloucester, MA 01930, US
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Microphones, Haptics, and the Metaverse

This February edition of *audioXpress* always involves working during unusual circumstances, when it looks like half the world is just thinking about the holidays and end of the year festivities, while within our team and our industry circles we are struggling with anticipated deadlines and planning for CES—trying to determine an agenda that will reward us with maximized returns for the following year. I know this will likely sound bizarre considering the timing as you are reading this. But I'm leaving these thoughts here mainly for future memory—and I know I will enjoy reading about it, and looking back at these unusual moments.

After all, the content focus for this edition are microphones and capturing audio and voice at the source—which is something that made more sense back when we thought this would be the issue distributed at the Winter NAMM Show. Of course, for 2022 the NAMM association has already announced that the NAMM Show will take place June 3–5 instead—and that's probably going to be one of the first major audio trade shows to return, if all goes well. In fact, the NAMM should be praised for anticipating the pandemic challenges. While the audio industry overall continued to grow throughout the pandemic, a more detailed look at specific application segments, such as musical instruments and live sound, doesn't paint such an optimistic picture at this time of writing. Right now, extreme measures, such as planning a show to take place six months later than its traditional schedule out of abundance of caution, seems a noble thing to do.

The NAMM Show is the annual "family reunion" of the global music, sound, and entertainment technology industry, and the decision to reconvene in Anaheim, CA, in June 2022 makes sense from a public safety standpoint and was met with support and enthusiasm across the industry.

The unpredictability of the industry under these circumstances also means that, while we try to remain as connected as possible to some sort of reality, we've all been virtually connected for two full years now. And not surprisingly, this is having long-lasting effects on technology and what we all prioritize as consumers. Among the technology buzzwords of the moment is the absurd notion of "metaverse," a clear result of pandemic-fatigue.

I believe remote collaboration platforms will evolve with dedicated and unique features, and that we will soon see Augmented Audio and Augmented Reality applications to feel excited about. Technologies that will augment OUR reality—not a dystopian second-life universe with humans connected by virtual reality (VR) goggles. And I believe that the audio industry, and very specifically microphones and voice front-ends, will play a key role in this (for now utopian) concept.

As I wrote in *The Audio Voice* newsletter, I see Audio Augmented Reality (including "super-hearing" capabilities) and language translation as very promising applications for the related technologies. Too much money has already been invested in VR device industries that would be better allocated in augmented audio.

Some notable progress is described in this edition of *audioXpress*, discussing Haptic Devices for Augmentation, Entertainment, and Wellness (a key part of the metaverse, if you care about that), microphone technology, and machine learning contributions for voice recognition. We even intended to expand this edition with an overview of voice front-end solutions and MEMS microphones, but we realized that the intended scope was already too ambitious to be featured in this issue. Therefore, we will be addressing those topics in the March issue, which (believe it or not) we are writing before the end of 2021.

For the April 2022 edition of *audioXpress,* we will be expanding the focus to the exciting field of true wireless earphones and hearables, where active noise cancellation, voice interfaces, and real-time adaptive processing are propelling the microphone market to new heights.

I know there are already many developers and builders working in this domain, with some of them making their efforts available to the open source community. I believe it's time for some DIY contributions in this field and *audioXpress* is looking forward to publishing those articles. Please contact editor@audioxpress.com

J. Martins
Editor-in-Chief

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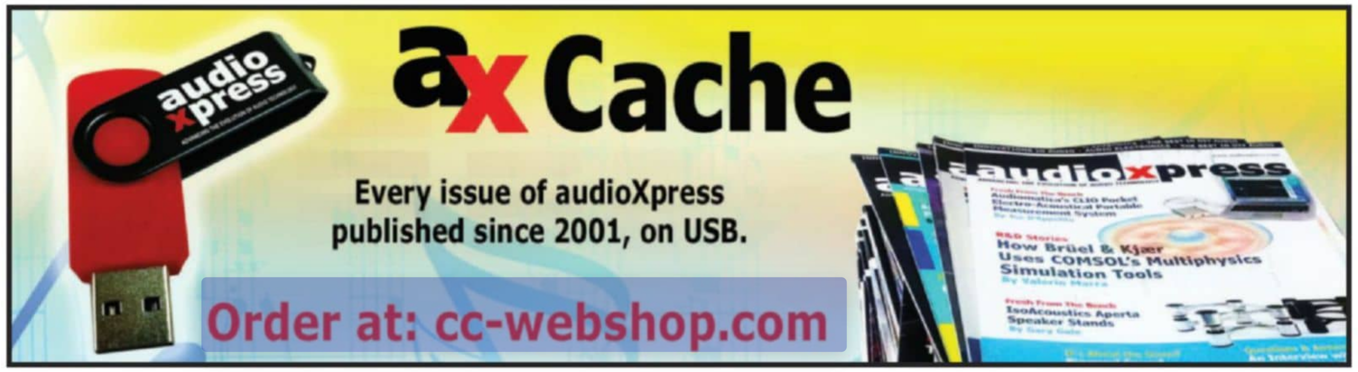

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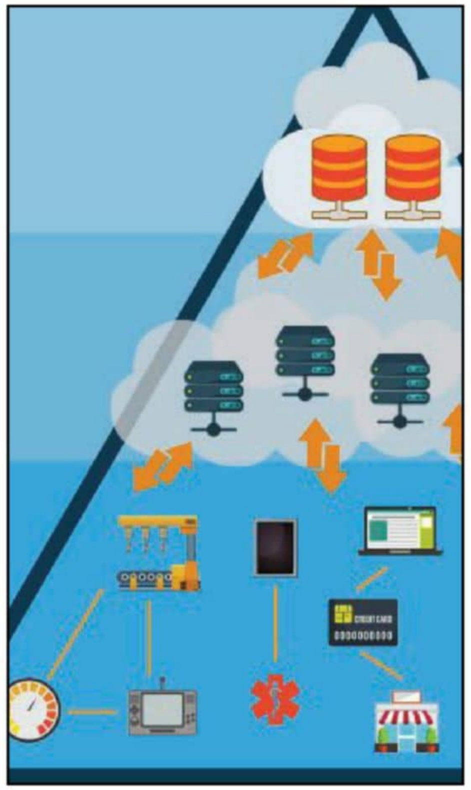

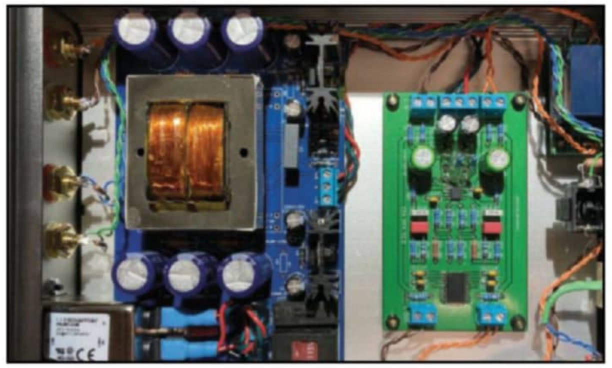

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The Evolving Microphone Market
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Market Update—Microphones

From XLR to USB

The Evolving Microphone Market

By
J. Martins

Our annual update of microphone technology addresses the latest products and trends in microphones. Traditionally a market targeting musicians, podcasters, and creatives—product segments that all continue to see strong demand—microphones are increasingly used to capture voice and our surroundings, and becoming an essential component in all audio product designs and our daily lives.

The beyerdynamic M 70 PRO X microphone is optimized specifically for speech and podcasting applications.

Many companies have entered the microphone segment in the past two years to explore the best approach to support remote work with efficient communications. With the pandemic effects extending throughout 2021, any audio company with a connection to the music world or recording world has also entered the microphone segment, in most cases with cheap outsourced products that are bundled with something else.

At the same time, many technology and consumer electronics companies sense that microphones are a changing environment, with new technologies allowing new use cases and are exploring completely new approaches. We see versatile microphones that are designed to connect with cameras or mobile devices that are far more versatile for content creation. And we see companies exploring the thriving gaming, streaming, and podcasting application areas to optimize “direct-to-computer” designs. And microphones are increasingly being used always-on, as sound sensors that can be quickly adjusted to different applications. The advancement in digital interfaces and smart-amp ICs allows new classes of microphones that can be configured remotely via software. At the same time, miniaturized microphones are everywhere and they are being used for many new applications, from voice interfaces to acoustic compensation—and MEMS microphones are getting much better and evolved for those applications. This is a thriving environment that continues to make microphones an always exciting category.

Confirmation of Trends

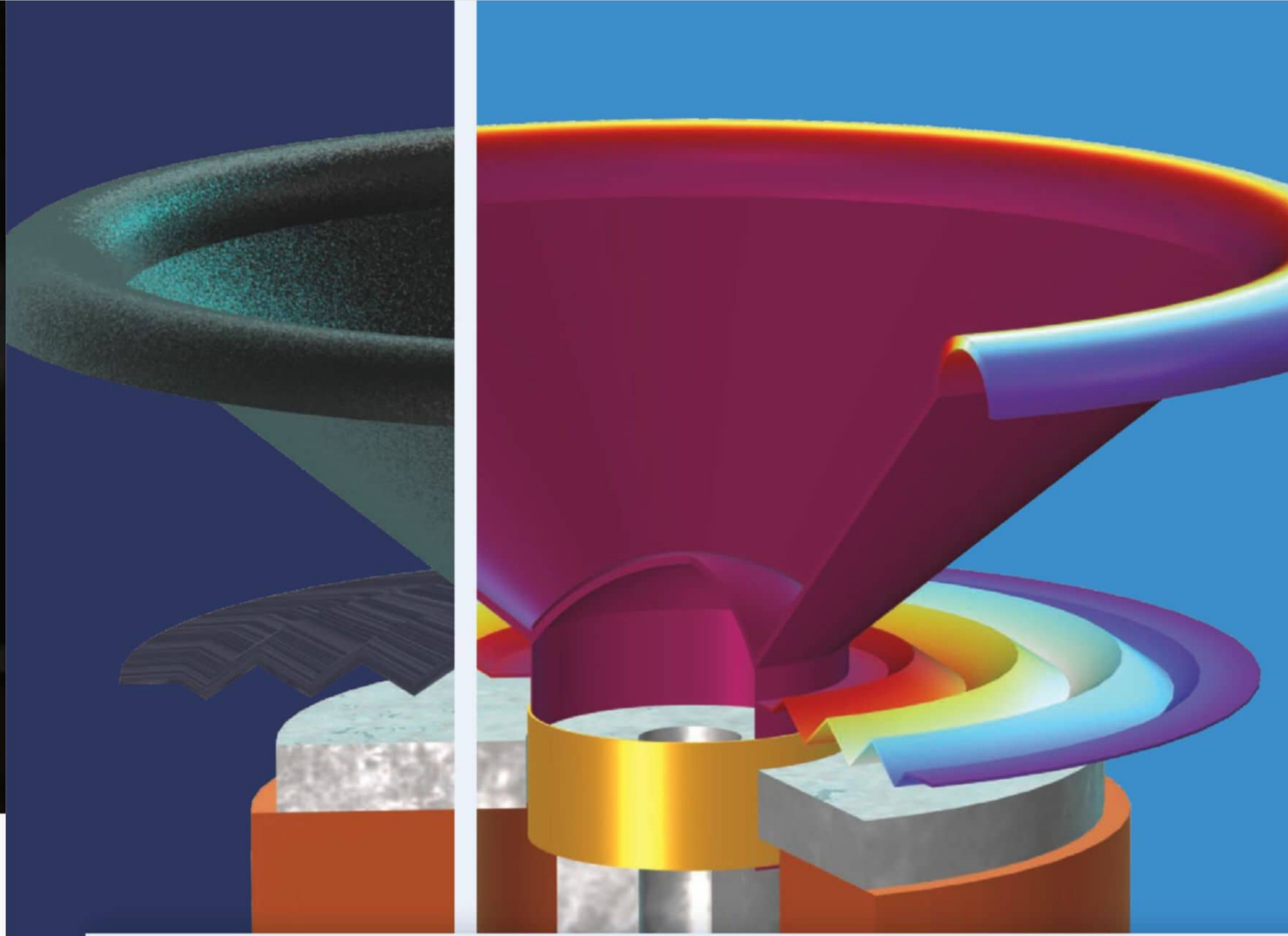
The year of 2021 has completely accelerated and confirmed the trends we reported in our previous Market Update on microphones. As the world converted work-from-home and work-from-anywhere spaces into recording and live streaming studios, activity among traditional microphone manufacturers and many new audio and technology companies continued to flourish with many novel products.

As the industry continues to explore the best approach to support remote work and respond to new requirements and use cases, we have seen major announcements from both professional manufacturers, consumer electronics, and gaming brands that are responding in innovative ways. Meanwhile, some major players are aggressively expanding their market reach by exploring a hybrid hardware-software approach, while companies active in the studio and recording markets are pushing the bold concept of microphone response emulation even further.

These are all trends we previously noted in our February 2020 report, as the effects of the global pandemic were starting to be felt, and in February 2021, when *audioXpress* highlighted how the microphone category was quickly changing.

The microphone market continues to grow consistently, with the exception of live touring and stage application segments, where professional wireless microphones systems are still not selling anything close to the record levels seen in 2019. Every other category is seeing significant growth in volume units, even if for the larger and most traditional microphone brands, the business volumes have not been sufficient to offset the effects of stagnant sales in the pro audio segment.

According to Austrian Audio, the relatively new microphone company that inherited the long tradition of AKG in the segment, the market has been challenging for a new pro audio player that



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One of the few professional wireless microphone system announcements in 2021, Sennheiser introduced the Evolution Wireless Digital microphone transmitters complemented with an app-based guiding system to make wireless easy and reliable for corporations or schools.

began activities before the pandemic and was just establishing its global distribution network. Still, the company reported “record-breaking sales” in 2021 and released several new microphones and headphones this year. As Martin Seidl, the CEO of Austrian Audio, explains: “Even though the Covid-19 pandemic made the live music market unpredictable, Austrian Audio leadership decided to move forward and create products for the stage as well as the studio.” The company just started shipping its new true condenser vocal microphone, the CC8, a hand-built cardioid true condenser microphone using its OCC7 small-diaphragm condenser capsule, reinforcing its credentials among professional users.

Studio microphones, for professional recording or home and project studio recordings, are seeing increasing volume numbers, but the real important trend is the growth in new categories related to live streaming, remote work, communications, podcasting, and even mobile capture. And users seem to be increasingly aware of wireless convenience. In fact, wireless microphone applications using consumer level technologies, such as Bluetooth, are on the rise and will be one of the most promising segments for the future.

Austrian Audio OD505 Active Dynamic handheld vocal microphone are designed for live performance, featuring a design that minimizes the contact between the capsule and the enclosure.

Modeling Microphones

Also as noted in our previous Market Update, the trend for “virtual microphones” being adopted in the studio continues to accelerate, with Antelope, Townsend Labs, Slate Digital, and others gaining the market as more musicians and producers recognize the virtues of microphone modeling.

In line with this trend, one of the most significant stories in the microphone market for 2021, was the acquisition by Universal Audio (UA) of Townsend Labs, the microphone modeling pioneer behind the TEC award-winning Sphere L22 Microphone System, introduced in 2016. With the studio market seeing increasing acceptance of microphone modeling technology, the Townsend Labs Sphere L22 microphone system now joins UA’s industry-leading line of audio recording products, including Apollo interfaces and UAD Powered Plug-Ins. Powered by proprietary microphone modeling technology, the Sphere L22 mic system provides a wide range of highly sought-after studio mic sounds with amazing accuracy, allowing for auditioning and selection of various mic types and polar patterns, even after audio has been recorded.

Townsend Labs was founded in 2015 by 13-year Avid alum and DSP architect Chris Townsend, together with Erik Papp, previously the CEO of Summit Audio. The Sphere L22 microphone was launched with a phenomenally successful international crowdfunding campaign on Indiegogo, raising 250% of its target in the first 24 hours. The high-precision, side-address, large-diaphragm condenser microphone, combined with microphone modeling software clearly made sense to the market, emulating the characteristics of the most sought-after microphones.

UA has partnered with Townsend Labs since 2017, most notably on the Ocean Way and Bill Putnam Microphone Model Collections for Sphere L22, directly supported by UAD and Apollo DSP platforms. UA says it will continue to support all of the Sphere L22 system’s current audio platforms, including UAD, AAX DSP, AAX Native, VST, and AU. But it will be interesting to see what the company will do with the technology. UA just recently hired Sandeep Gupta (Digidesign, Apple, and Amazon) to be the new Chief Operating Officer and accelerate its growth. With this agreement, Townsend Labs’ founders Chris Townsend and Erik Papp have also joined the UA team, working to develop future microphone products under the UA brand.

In 2021 Universal Audio acquired Townsend Labs, the microphone modeling pioneer behind the TEC award-winning Sphere L22 Microphone System.

And since we are on the topic of modeling microphones, the product announcement of the year on that front was the new Axino Synergy Core from Antelope Audio, a new design that combines a cardioid condenser modeling microphone and a USB audio interface.

The company was the first in the market to introduce a USB-C bus-powered modeling microphone, enabling a complete studio experience with circuit-level microphone and dynamics emulations onboard Antelope Audio’s real-time FPGA FX platform, back in 2019. Before that, professional audio engineers tended to dismiss USB microphones as cheap, unimpressive products for beginners—and in many cases, they still have a point. But Antelope made it possible to have the full, non-compromised fidelity of a large-diaphragm condenser microphone, powered by a console-grade mic preamp, high-end AD converter, and signature processing, leveraging the power of an USB-C interface.

The new compact Axino Synergy Core microphone is an all-in-one studio-grade recording system, anchored around the company’s microphone Modeling Engine, proprietary Synergy Core effects platform, and 24-bit/192kHz AD/DA conversion. A versatile tool for recording, starting with a good-quality microphone, featuring a large, gold-spluttered diaphragm that delivers low self-noise. The cardioid pick-up pattern minimizes background noise capture in favor of focusing on the sound source directly in front of the mic itself. And built inside this studio-grade microphone is also a preamp that is a hybrid of the renowned discrete preamps Antelope Audio uses in its professional audio interfaces.

But the new Axino Synergy Core was designed for microphone modeling, which Antelope Audio has been successfully exploring in its most recent generation of studio products. At a very affordable price, it enables access to 18 emulations of legendary microphones, which can be applied in real time during recording or when mixing in Digital Audio Workstation (DAW). Best of all, users can experiment with the flexibility and versatility on offer by choosing the microphone that best suits a particular project’s needs.

Even before opening the easy-to-use software control panel for macOS and Windows, the microphone itself offers volume and headphone controls alongside a -10dB pad switch and a high-pass filter switch, which helps with taming the original sound source. After that analog input stage, the Axino Synergy Core effectively offers an entire audio interface built into the studio-grade microphone, with powerful DSP and FPGA processing, precise clocking, and superior AD/DA conversion. According to Antelope Audio, using the same AFC technology from the company’s rackmount audio interfaces—a cutting-edge combination of clocking and conversion built into a USB microphone.

The Axino Synergy Core also comes with custom-built audio drivers for macOS and Windows, optimized for streaming and playback, as well as low-latency recording, while the software control panel permits easy access to settings configuration, signal routing, and effects processing.

Antelope Audio Axino Synergy Core cardioid condenser modeling microphone and USB audio interface

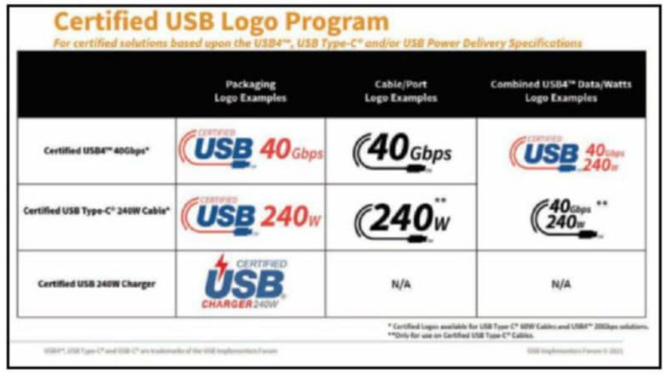
The New Microphone Interface

Probably the most important trend to take note this year—since it is almost transversal to all the application segments mentioned—is the maturity and dominance of USB microphones.

Previous discarded as an entry-level proposition for amateurs and gaming, USB microphones are now available in all capsule sizes and are starting to gain trust and even be preferred for home use and podcasting. If previously the argument of the additional gain awarded by a good quality microphone preamplifier was by itself a reason to choose standard XLR microphones, the confidence awarded by mobile applications where microphones are powered by battery-operated mobile devices, and the availability of higher power USB specifications, is paving the way for new “pro-oriented” microphone designs that advantageously rely on USB interfaces.

More important, the evolution to USB Type-C and Power Delivery, enabling computers and mobile devices to be able to power and recognize USB microphones and integrated circuits, paved the way for interesting new software tools that add to the fundamental technology. And we are starting to see examples of companies that are using software processing to build upon USB microphones by adding innovative features: the emulation of responses, microphone arrays, channel groupings, and the ability to connect multiple microphones for streaming or recording, all are appealing propositions.

In fact, what we are starting to see happen in innovative USB microphone designs clearly indicates that very soon we could see



New Certified USB Type-C Cable power rating logos highlight support for 60W or 240W of power, as defined by the USB Power Delivery (USB PD) 3.1 Specification.

Recording an acoustic guitar using the improved microphone system in Apple's latest MacBook Pro M1 Pro. (Image courtesy of Andrew Huang's YouTube video)

this interface as superior to the standard analog connectivity and external preamplifier.

With the introduction by the USB Implementers Forum (USB-IF) of Certified USB Type-C Cable power rating logos to highlight support for 60W or 240W of power, as defined by the USB Power Delivery (USB PD) 3.1 Specification, that perception will increase among professionals, which until now pretty much ignore USB as an option for microphones. The latest Certified USB4 logos have even been updated to unify branding across the Certified USB Logo Program, offering consumers a simple way to identify the performance and power capabilities of certified USB cables, hosts, devices, and chargers.

Previously, USB Power Delivery (PD) was limited to 100W via a solution based on 20V using USB Type-C cables rated at 5A. The USB Type-C specification has also been updated with Release 2.1 to define 240W cable requirements, and with the updated USB PD protocol and power supply definition, this extends the applicability of USB PD to a large number of applications where 100W wasn't adequate.

EPOS' Adapt 200 Bluetooth headset and B20 USB-C microphone have features directly inspired by the gaming market.

The USB PD 3.1 specification describes a choice of three new fixed voltages: 28V (above 100W), 36V (above 140W) and 48V (above 180W) joining previously defined 5V, 9V, 15V, and 20V fixed voltages. This means that supplying power to large condenser microphones, large TVs and displays, or powering large network setups from a single USB-C adapter converted to PoE becomes a possibility. For audio applications, particularly for professional applications, the new adjustable voltage mode, enabling a range from 15V to one of three maximum voltages (28V, 36V, or 48V) depending on the available power, allows the device being powered to request specific voltages to a 100mV resolution.

The new USB Power Delivery architecture also defines a much more stringent power negotiation protocol that helps to ensure that access to and use of this higher power capability can be done safely.

These updates are not only going to impact microphone applications but professional audio equipment in general, starting with USB-powered audio interfaces and mixing consoles.

To mention a recent example, Universal Audio (UA) surprised with the introduction in October 2021 of a completely new range of USB-C audio interfaces, with five different models all offering best-in-class sound quality, with built-in DSP. These are designed to power traditional XLR microphones and high-impedance audio sources such as guitar pickups.

UA is a company with great traditions in audio hardware and more specifically in studio audio interfaces that always have combined the most advanced choices of physical data interfaces (Firewire, Thunderbolt, etc.) with built-in DSP power to create fully integrated solutions that offer the best of both worlds, including unique software emulations of analog circuits that become accessible in real time and without latency.

Now, thanks to the latest updates to the USB specifications and the introduction of USB Type-C connectors, USB audio offers the perfect combination for project and home studios, and we wouldn't be surprise to see solutions evolving to support USB audio peripherals in general—microphones being the most obvious.

Small and Miniature Microphones

Microphone technology is also evolving fast in the miniature, low-power category of components, with MEMS microphones being pushed further due to integration in mobile devices, laptops, tablets, and even appliances to serve all common users-needs in terms of interface, communications, and even audio recording.

A great example can be found in the most recent MacBook Pro M1 laptops from Apple—the 14" and 16" models—which not only deliver groundbreaking processing, and graphics performance, an improved headphone jack that supports high-impedance headphones, and the best sounding six-speaker sound system in a laptop ever, but are also extremely good in the microphone department.

According to Apple, not only do the new 2021 MacBook Pros have "the best audio system" and "the longest battery life ever" in a notebook, the microphone system was designed to enhance any kind of audio source—not just FaceTime calls. The new MacBook Pro M1s have microphones with an even lower noise floor, resulting in clearer calls and voice recordings. Not surprisingly, some users decided to test the results with acoustic guitars and singing and were very positively surprised. After some physical adjustments to change the position of the laptops because of the microphone beamforming—including placing the laptops sideways on a desk—the results can rival larger membrane microphones specifically designed for home studio recording (check out this interesting video by Andrew Huang: https://youtu.be/PL6XBdYxOtE).

That is also because subminiature, piezo, and MEMS microphones are all being pushed further because of voice recognition and demanding applications in smartphones and wearables (of course, combined with very powerful DSP processing) that they are able to rival standard electret condenser microphones (ECMs) in most conventional applications—and beyond, as we can see in Apple's products.

According to French market research firm Yole Développement, which recently published an updated 2021 study on consumer MEMS microphones, the segment will reach approximately $2.5 billion USD in 2026 at 10.5% compound annual growth rate (CAGR), as microphones are increasingly being integrated into all types of consumer devices to support smart assistant integration, noise cancellation, and immersive audio experiences. Microphones, together with other audio components, are also making their way into cars in several applications, from voice interaction to noise cancellation and supporting virtual acoustics. But that is a topic *audioXpress* will address in a separate article.

Hybrid Working

The other main trend in the microphone segment is obviously related to the evolving landscape of remote home work, work-from-anywhere, and hybrid-working environments. Many companies previously focused exclusively on the unified communications and office market—with products that were intended for business executives and call centers alike—have quickly adapted in 2020 to try to meet the demand from workers forced to remain at home.

In 2021, these same companies have understood that, in order to compete, they would need to quickly embrace the technology trends happening in the consumer space—as many remote workers established Apple AirPods for that effect with considerable success. And with ANC now well appreciated in consumer headsets, it also became a must-have on the return to open offices and shared workspaces.

When workers started to face returning to offices and resuming travel, companies such as Poly (resulting from the Plantronics/Polycom merger), Jabra, and Logitech all felt the need to reevaluate their designs in order to anticipate the market's needs. One of the most dynamic companies in this space is Danish innovator EPOS (previously in the market as Sennheiser Communications), which commissioned a research study about the challenges of remote work and the use of technology at the heart of hybrid working.

The data collected in early 2021 for EPOS came from a target sample pool of more than 2,500 male and female respondents from the US, UK, France, Germany, Hong Kong, and Singapore, and included respondents working in companies with 50+ employees, decision makers, and end-users of audio solutions.

The results have shown that business leaders and employees are now demanding audio and video solutions to help maintain

HyperX QuadCast, Quadcast S, and SoloCast USB microphones are now all part of HP's gaming division.

standards, while benefitting from a mix of remote and office-based working.

More important, they all see hybrid working as the future, and want personal solutions that they can take home and everywhere they go. The concept of "office equipment" that you just leave on the office desk is no longer seen as compatible with safety and hygiene practices. Employees expect companies to provide them with products that are as good as or better than those they would buy from themselves, wireless, fast-charging, and ANC capable. Those businesses that embrace a thoughtful and strategic approach to the future, considering the views of their workforce, are likely to thrive and attract the best talent.

"Organizations can choose to invest in high-quality, engineered audio solutions to meet the demands of on-the-go work. And those that do may edge the competition in the post-pandemic world," says Jeppe Dalberg-Larsen, President at EPOS.

With the right technology solutions, EPOS found that 64% of end users say that quality audio solutions enable them to capture critical information, while 61% believe the right solutions empowers them to appear professional. Employee wellbeing is tied to this and is critical to the success of any organization. Technology designed for the office environment often doesn't work for a remote setup and that is the reason why EPOS is quickly renovating its product portfolio

The new Neumann KK 14 capsule is the core element in the Miniature Clip Mic System, with nine options for reliable attachment to brass, string, and stringed instruments, pianos, drums and percussion.

with the latest technologies, focusing on high-end audio. Typical miniature microphones used in office headsets are being replaced with microphone arrays that are also able to support noise reduction and ANC, or even voice recognition. And the company's latest Adapt headsets directly target the hybrid workforce, with a noise cancelling microphone to augment the conversation experience, travel-friendly folding designs, Bluetooth, and even support for different types of connectivity, including either a USB-A or USB-C dongle for old PCs without Bluetooth.

And EPOS is also using lessons learned from younger users and its own gaming product range, and offering multimedia options for the desktop. That is the reason why EPOS has also decided to enter the thriving USB-C desktop microphone segment in 2021, and launched the B20 broadcast microphone ($199.00 USD). While being marketed as a gaming product, the B20 is also targeted at streamers and content creators seeking studio-quality microphone audio. Using three condenser capsules, this USB-C design is compatible with PC and Mac computers, as well as PS4 consoles, delivering 24-bit/48kHz audio, and enables users to select between four different polar responses—cardioid, stereo, bidirectional, and omnidirectional—supporting multiple application scenarios.

The B20 microphone also includes integrated audio controls for volume, gain, and mute and includes a headphone jack for real-time voice monitoring and game audio mixing. With a refined aluminum design, it ships with a dedicated desk stand that allows for perfect fit and adjustability, while a 3/8" thread seamlessly connects to boom arms for a range of added positioning options.

The hybrid working/gaming environments could also explain why, in February 2021, HP decided to acquire HyperX, the gaming division of Kingston. Just 8 months later, we learned that HyperX had surpassed one million USB microphones sold, exceeding all of the company's expectations for the category. Initially introduced into the microphone market in 2019, the HyperX QuadCast USB microphone became a popular choice for gamers, streamers and influencers. As demand for microphones increased, HyperX expanded its portfolio with the customizable QuadCast S and affordable SoloCast to fulfill the need for gamers, content creators, students, and business professionals working from home.

DPA Microphones has the 4099 CORE Instrument Microphone and an extensive range of accessories for all musical instrument applications.

HyperX microphones feature a unique, easy-to-use tap-to-mute sensor with LED status indicator, offering users a visual LED lighting cue when the microphone is active. These versatile and affordable microphones also fit 3/8" and 5/8" threaded setups for compatibility with most mic stands and boom arms. And all are compatible with Windows PCs, Macs, and PS4, and they work on major streaming platforms (e.g., Streamlabs OBS, OBS Studio, and XSplit) and are certified by Discord and TeamSpeak.

HyperX QuadCast and Quadcast S microphones offer four polar patterns—stereo, omnidirectional, cardioid, and bidirectional—to support a variety of broadcast setups and applications, and come equipped with a built-in anti-vibration shock mount and pop filter, conveniently located gain control dial and a 3.5mm headphone output for live mic monitoring.

The HyperX SoloCast USB microphone leverages a single condenser design with a rotatable stand and adjustable tilt to fit in small, tight spaces such as under a monitor. This microphone uses a cardioid polar pattern that prioritizes sound sources directly in front of the microphone to deliver quality audio for video editors, streamers, gamers, and more.

Modular Pro Microphones

All the trends described in this introduction are better illustrated with the multiple examples of new products introduced in the past 12 months, and we can say the year was rich in great microphone designs.

As an example, just as we were wrapping up this Market Update, we received note of a new microphone launch from Neumann, and more interestingly, a totally new category for the brand. In fact, Neumann.Berlin introduced a whole product range designed especially for close miking musical instruments. The new modular Miniature Clip Mic System is based on a new KK 14 electret capsule that Neumann's developers spent six years researching, together with the production processes that would be necessary to make this a reality.

To create its new Miniature Clip Mic System, Neumann focused on stability and long service life for live and stage performances. The system includes nine mounting solutions that make it possible to find the optimum positioning for acoustics while simultaneously protecting the instrument. The most important thing, naturally, is the sound delivered by the Neumann KK 14, which the Neumann engineers in Berlin define as "unlike anything that came before."

With minimized self-noise and an extremely high maximum sound level of 152dB, the new Neumann KK 14 should deliver perfect results for both very quiet and very loud instruments. Neumann assembles the KK 14 capsules by hand and encapsulates them in a titanium housing. The capsule can be easily unscrewed from the gooseneck connector whenever necessary. All the other components are also easy to replace. A combination of factors that Neumann believes offers a complete microphone system with a superior return on investment.

This new Neumann microphone directly competes in an area with multiple established brands, including DPA Microphones, the Danish microphone manufacturer that in 2021 saw its renowned DPA 4006 microphone capsules being used by NASA in its Mars Perseverance Rover.

For this historic mission, the rover was outfitted with a DPA system, consisting of a 4006 Omnidirectional Microphone, MMA-A Digital Audio Interface, and MMP-G Modular Active Cable, to provide sounds from the surface of Mars. After the rover touched down on the Red Planet, audio and video files were recorded from the surface and transmitted to NASA's base station.

This is a testimony of the rigorous build and qualities of DPA's microphones, which are also a professional choice in musical instruments capture, particularly since the introduction of the 4099 CORE Instrument Microphone condenser range and accessories. Available in "Loud SPL" and "Extreme SPL" versions, these microphones excel at rejecting unwanted noise and accurately capturing the true voice of multiple acoustic instruments.

There is also a Knowles EK series microphone in the SuperCam aboard the rover—selected for its durability, high acoustic sensitivity, and low vibration sensitivity needed to transmit sound from the Martian atmosphere. This one was able to pick up the sound of the Ingenuity Mars Helicopter on its fourth flight, on April 30, 2021.

And DPA continued to expand its latest microphone solutions, designed to meet the growing production needs of the broadcast and performance markets. As the rapid adoption of live streaming continues to grow in broadcast applications, speech intelligibility remains at the forefront of all productions. In recognition of these changing requirements, DPA recently released the 4466 CORE Omnidirectional and 4488 CORE Directional Headset Microphones, and the 4097 Micro Shotgun Microphone and Interview Kit.

Based on the design of the company's groundbreaking 6066 Subminiature Headsets, the new 4466 CORE Omnidirectional and 4488 CORE Directional Headset Microphones feature the brand's popular 5mm round microphone capsule. With a one-size-fits-all design and adjustable boom height and length, the 4466 and 4488 headsets accommodate all head shapes and sizes. A unique, three-point gripping system (above, below, and behind the ear) ensures a greater level of security, while the flexible ear hooks provide continued comfort during extended use.

Featuring the same sonic qualities as the brand's supercardioid Choir Microphone, the 4097 Micro Shotgun Microphone has been optimized to pick up speech from a distance—ideal for use as part of the company's new 4097 CORE Interview Kit, which includes a lightweight boom and windjammer, and allows journalists to conduct interviews from up to 6.5' (2 meters) away, for single-reporter interview applications in today's socially distanced world.

New Notable Microphones

Delivering on the promise of significantly expanding its product portfolio for demanding recording applications, in 2021 Audix launched the new A127 Omnidirectional Metal Film Condenser Microphone. For the Oregon-based microphone manufacturer, the A127 is the culmination of years of research and development in condenser technology.

The Audix A127 Omnidirectional Metal Film Condenser Microphone is the culmination of years of research and development for the Oregon-based microphone manufacturer.

The Audix A127 omnidirectional microphone features a reference-grade, pre-polarized half-inch Type 1 metal film capsule (IEC 61094-4 compliant), that delivers the critical audio capture quality required for professional studio recording, broadcast, film, and sound design. With a frequency range between 10Hz to 20kHz, 133dB of dynamics, and an ultra-low self-noise of 7dBA, the A127 excels in the most demanding applications.

The capsule design features a precision-tuned, 3-micron metal diaphragm, and a circuit designed and built to exacting specifications, coupled to the capsule by a series of driven shields to protect the microphone from external interference and capacitive coupling. The mechanical components of the gold-plated shield and the corrosion-resistant housing are all precision-machined by Audix in Wilsonville, OR.

From concept to completion, Audix's onsite research and development team, combined with an in-house manufacturing facility, was able to complete the A127 project, which represents the culmination of years of R&D in condenser technology.

Another major announcement from Audix centered on the A231, a large-diaphragm studio vocal microphone that the company designed to open up new creative possibilities in the realm of lead and backing vocals, spoken word and voiceover applications, as well as acoustic instrument miking.

Another major announcement from Audix in 2021 was the A231 large-diaphragm studio vocal microphone.

"Clean and powerful" are the adjectives chosen by Audix to characterize the new A231 condenser microphone, which also stands out for its warm character, and low self-noise. Bringing together multiple high-end design features, the gold-plated brass band around the circumference of the A231 hints at the revolutionary capsule within: a 33mm, true condenser featuring a precision-tensioned diaphragm with gold deposition all the way to its far edges. This produces an effective diaphragm area approximately 25% larger than that of a typical large-diaphragm condenser mic, and results in a sound that is neutral and rich at the same time. An exacting cardioid pickup pattern offers a predictable proximity effect and exceptional off-axis rejection.

Austrian Audio's new CC8 small diaphragm microphone features the company's new OCC7 condenser capsule, inspired by the legendary AKG CK1, updated to meet the demands of today.

The capsule's circuitry and signal path is protected from interference and capacitive coupling by an active driven shield. An internal shock mount virtually eliminates touch noise, enabling the A231 to be used across a wide variety of environments, including a live stage—without the need for an unwieldy external shock mount.

The A231 capsule is suspended by a specially designed internal shock mount behind a double-woven steel alloy mesh grille. The elegant, yet compact, precision-machined aluminum body was also designed to be practical—allowing the A231 to be optimally positioned in tight spaces where larger microphones may be too obtrusive.

As is the case with many premium vocal mics, with a massive 140dB maximum SPL, the Audix A231 is also fit to capture acoustic instruments and can easily handle a kick drum at close range, yet its low 12dBA equivalent noise level makes it whisper-quiet on more delicate sources (e.g., an upright bass or strings).

Another busy company in 2021 was Austrian Audio, which in August unveiled two new microphones that expand the company's product catalog onto the stage. The new OD505 Active Dynamic and OC707 Handheld Condenser are handheld vocal microphones that are designed for live performance with capsules that are handmade in the company's facilities in Vienna, and a design that minimizes the contact between the capsule and the enclosure.

The new OC707 is a rugged vocal microphone with a die-cast body and true condenser technology, featuring a small-diaphragm condenser capsule with a frequency response optimized for the human voice. Its supercardioid polar pattern with excellent off-axis rejection ensures maximum suppression of feedback. The combination of low self-noise (19dB SPL) and high sound pressure level (150dB SPL) results in an impressive dynamic range, creating a vocal microphone that has all the important characteristics of a studio mic but can withstand the rigors of everyday stage use.

The OD505 is an Active Dynamic microphone that combines the advantages of a dynamic stage mic with the sonic refinements of a condenser microphone. This cardioid vocal microphone achieves nearly the same sensitivity as a condenser mic due to its state-of-the-art dual capsule design where the bottom capsule is phase-

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AKG | www.akg.com

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Audio-Technica | www.audio-technica.com

Audix | www.audixusa.com

Austrian Audio | www.austrian.audio

beyerdynamic | www.beyerdynamic.com

DPA Microphones, Inc. | www.dpamicrophones.com

Earthworks Audio | www.earthworksaudio.com

EPOS | www.eposaudio.com

HP Development Co. | www.hp.com

HyperX | www.hyperxgaming.com

Mackie | www.mackie.com

Neat Microphones | www.neatmic.com

Neumann.Berlin | www.neumann.com

Røde Microphones | www.rode.com

sE Electronics International, Inc. | www.seelectronics.com

Sennheiser electronic GmbH & Co. KG|USA | www.sennheiser.com

Turtle Beach | www.turtlebeach.com

Universal Audio | www.uaudio.com

Analog Gain Stage

The THAT1580 and THAT1583 are very low noise pre-amplifiers with EIN figures of -134.8dBu and -128.9dBu, respectively. They are super fast (>50V/μs) and high voltage. Overall, the best start to your signal chain.

Digital Controller

The THAT5263, THAT5173 and THAT5171 are digital gain controllers that add convenience and functionality. They make it easy to optimize signal to noise at the critical first gain stage before digitization.

All-in-One Solution

The THAT6261, THAT6263 and THAT6266 include both the pre-amp and digital controller in one small QFN package. What's more, they use only 30mW of power per channel.

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inverted and passively switched. Combined with a switchable second-order high-pass filter and the company's Open Acoustics technology, noise from handling, bumps, and stage rumble are effectively eliminated.

Both new microphones utilize Austrian Audio's special Open Acoustics technology, which allows sound to enter the capsule almost entirely unimpeded from all sides. Contact points between the mounted capsule and the enclosure are reduced to a minimum, allowing for an open and natural sound without any coloration.

Another major announcement for Austrian Audio in 2021 was the first of a new line of instrumental microphones for use in the studio and on stage. The new CC8 small diaphragm microphone features Austrian Audio's OCC7 condenser capsule and was inspired by the legendary AKG CK1 introduced in 1969.

The capsule's three-micron-thick gold-coated polyethylene napthalate (PEN) diaphragm is much more robust than conventional mylar alternatives. The OCC7's filter fabric guarantees a constant broadband cancellation, resulting in cardioid characteristics with exceptional attenuation of sound coming from the sides and rear.

The capsule and transformerless output give the CC8 an extremely linear frequency response. It can handle extremely high sound pressure levels of up to 156dB SPL without distorting. Combined with a very low self-noise of only 16dB SPL (A), this results in an exceptional dynamic range. Excellent for recordings in which fast transients and spatial information needs to be captured as accurately as possible.

And thanks to the built-in switchable attenuation of 0dB, -10dB, and -20dB, the microphone can be positioned very close to loud sound sources (e.g., trumpets) without fear of distortion. Rumble and impact noise are effectively eliminated with the switchable high-pass filter (60Hz second order, 120Hz second order).

Each microphone is measured and tuned in the company's in-house anechoic chamber to a maximum tolerance of only ±0.5dB sensitivity at 1kHz. This means that any CC8 can be match-paired with any other, regardless of their production dates.

The Røde Revolution

If there was a microphone brand that surprised and delighted with its new products and concepts, it has to be Røde Microphones. Leading in the category of microphones for mobile recording and content creation for many years, Røde remained faithful to its core market but managed to positively innovate and surprise in every single announcement—and there were many!

The latest announcement for Røde Microphones in 2021 was the new Røde Lavalier II microphone, which followed the introduction of the instantly successful AI-Micro ultra-compact interface with support for mobile recording from two microphones.

A new addition to its range of compact wearable microphones, the Røde Lavalier II is a premium lavalier microphone for the most demanding audio and video applications. The very low-profile design sets it apart from all other lavalier microphones and highlights the state-of-the-art capsule technology and unique form factor that allows the Lavalier II to sit flush against clothing and other mounting surfaces.

Specified as a broadcast-grade microphone with flat frequency response and omnidirectional polar pattern, the Lavalier II is ready to compete with the market references in that domain.

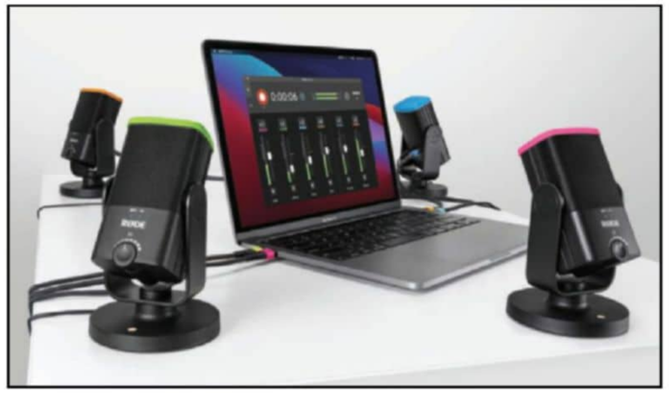
The Røde Lavalier II microphone is a premium lavalier microphone for the most demanding audio and video applications, showing strong ambitions from the Australian company.

Key features for this new lavalier design include an ultra-compact mounting clip—incredibly discreet and easy to conceal. It uses a simple 3.5mm TRS locking connector, which makes it compatible with a vast range of systems. And Røde also designed a premium accessory kit, which includes a pop filter, mini furry windshield, mounting clip, colored ID rings, and zip case, that increase the value of the new Røde Lavalier II.

The Røde Lavalier II is very forgiving when it comes to placement and pairs perfectly with the hugely popular Røde Wireless GO II transmitter/recorder and the newly released AI-Micro.

And if Røde leads the video capture market with its VideoMic range of directional microphones, the biggest success for the Australian company in the past two years has been the incredibly popular Rødecaster Pro Podcasting Console, which in itself unleashed a whole range of solutions, including great microphones. But not content with this technological accomplishment, Røde decided to take the concept further.

Inspired by the success of its Rødecaster Pro hardware console, Røde released Røde Connect, a simple and powerful software solution for podcasting and streaming with its NT-USB Mini microphone, acting as a fully featured recording interface. Not only that, but Røde introduced the ability to connect up to four NT-USB Minis to a single computer via USB-C, and with seamless integration with video call, streaming applications, and much more.



A revolution in the making. The Røde NT-USB Mini microphones, connected by the free Røde Connect software, enables up to four microphones connected to a single Windows or MacOS computer.

Røde Connect is a complete software solution for podcasting and streaming—the software is free and most of its features will work with any USB-connected sources, but its audio summing and processing abilities are directly unlocked by the Røde microphones that are connected. The free software serves up a fully featured recording interface, plus powerful routing options for live streaming applications. Røde Connect offers virtual channels for connecting remote guests, and automatic mix-minus—complete with broadcast-style faders, level metering, mute buttons, and more. An extremely powerful feature set, and also a powerful solution for gamers who want to simplify their audio setups by allowing control over microphone audio and external applications, plus dedicated output controls for streaming apps like OBS, TeamSpeak, or Xsplit.

Launched in 2020 as another USB microphone for all of us at home, the Røde NT-USB Mini is actually a studio-quality USB microphone, designed to deliver professional audio direct to a computer. A handy design for podcasters, streamers, musicians, and more. With Røde Connect running on both Windows and MacOS, the capabilities of the NT-USB Mini goes even further, with up to four NT-USB Minis connected to a single computer, and the ability to add more guests via Zoom and Skype, plus music beds, stings, and sound effects, pre-recorded interviews or any other audio from external sources.

Each NT-USB Mini microphone allows users access to a noise gate, compressor, and legendary Aphex Aural Exciter and Big Bottom effects, as found on the Rødecaster Pro hardware console.

Later in the year, Røde has released an update for Røde Connect, unlocking also compatibility with another popular Røde product, the VideoMic NTG shotgun microphone, and also the Wireless GO II microphone range. This allows this hybrid hardware/software solution to support all typical scenarios for content recording, from the typical roundtable interview for a podcast, to location interviews where a wireless lapel microphone is the more practical solution, and even a shotgun that will be more forgiving to distances.

And the versatile Røde Wireless GO II microphones can also be set up on the go with iOS and Android devices via the new Røde Central Mobile app, giving users complete control of their device.

This clever integration of products and continuously expanding range of accessories, make Røde Microphones a market reference in the segment, offering a new meaning to the ecosystem concept in content creation. And the company is only set to accelerate its portfolio in 2022.

Voice and Podcasting Microphones

A year after launching ICON, a studio-quality USB microphone, and ICON PRO, a broadcast-quality XLR streaming microphone targeting the thriving remote working, podcasting, gaming, and home recording application segments, Earthworks Audio launched ETHOS, a new condenser microphone tuned to broadcast standards. Once again fully developed in-house, ETHOS is a supercardioid 14mm condenser microphone with an extended frequency response that covers from 20Hz to 30kHz. It is able to handle extreme SPL levels up to 145dB, enabling even the most extreme of sports commentators to scream their hearts out. But its capsule was also designed to handle "the sound of relaxed confidence," as Earthworks describes the ETHOS.

The new microphone is obviously not targeted only at traditional radio or TV applications but will appeal in general to all voice and spoken word applications looking for a microphone that naturally enhances voice qualities. And Earthworks partnered once again with Triad-Orbit to design a microphone adapter, able to hold and balance the stainless-steel microphone body. An innovative windscreen stack with user-replaceable foam also allows the ETHOS to raise the bar for plosive and wind noise protection without impacting fidelity or transparency and ETHOS is naturally resistant to handling noise and thumps thanks to internal dampening and the robust stainless-steel construction.

From the expert hands of Audio-Technica, the AT2040 Hypercardioid Dynamic Podcast Microphone was another important introduction in 2021, taking inspiration from Audio-Technica's acclaimed BP40 Large-Diaphragm Dynamic Broadcast Microphone. The rugged AT2040 features an all-metal construction coupled with a new dynamic hypercardioid capsule that minimizes room noise, and features an internal shock mount and pop filter. This makes it a ready-to-use solution for home use and for working in less-than ideal acoustic environments. Not only for podcasting, but also for other content creation setups, such as live streaming, gaming, or doing voice-over work.

Directly aiming at the desktop space for content creators and streamers, Turtle Beach acquired Neat Microphones in January

The new Earthworks ETHOS supercardioid 14mm condenser microphone offers an extended frequency response to naturally enhance voice qualities.

Market Update—Microphones

Audio-Technica AT2040 Hypercardioid Dynamic Podcast Microphone is designed with an integrated shock mount to prevent unwanted noise and meet the needs of streamers, podcasters, and other content creators.

The Skyline USB desktop microphone by Neat Microphones offers high-quality, 24-bit/96kHz digital audio in an elegant and clean design, featuring a large mute button.

2021, one of those companies that was a victim of the Gibson Group debacle. The company was founded by Skipper Wise and Martins Saulespurens, also the co-founders of the successful Blue Microphones, later acquired by Logitech.

Already as part of Turtle Beach, the company launched the all-new Neat Skyline USB mic, the first product to be developed by the Neat Microphones team since the acquisition. A perfect desktop addition to improve sound capture when conferencing with co-workers, family, and friends, the Neat Skyline is a very affordable proposition with a really different design that stands out. And the Skyline offers high-quality, 24-bit/96kHz digital audio to any USB input and even includes a USB-C to USB-A cable for easy connection to an older computer.

In June 2021, Neat Microphones announced the King Bee II analog XLR microphone—the eagerly anticipated successor to Neat's original King Bee—offering improved versatility and sound quality for musicians, podcasters, or streamers, including spoken word applications, from voiceovers to podcasting to streaming content.

Later in the year, Neat Microphones launched the Bumblebee II Professional Cardioid USB Condenser Microphone, and the Worker Bee II XLR condenser microphone for high-SPL recording, both models improvement on the specs of earlier designs. The Bumblebee II features a larger, internally shock mounted 25mm capsule, making it a versatile and easy-to-use USB condenser microphone for content creators.

The Worker Bee II features a similar capsule, and targets instrument recording in home studios, with a tight pickup pattern and compact size that is perfect for positioning in small spaces and recording in treated and untreated rooms.

Another important new product launched in 2021 for this application space was the AKG Ara, a two-pattern USB condenser microphone that expands on the global success of Harman's AKG Lyra Multimode USB microphone, but more affordable. The AKG Ara follows the same design fundamentals of the Lyra, with a more streamlined format for two basic pickup patterns that make sense for tabletop use in webcasting and podcasts.

And beyerdynamic made an ambitious announcement already in October 2021, with its PRO X Series of headphones and microphones for creators. The headphones are the DT 700 PRO X and DT 900 PRO X, both very familiar designs that promise an optimized performance for studio work. The M 70 PRO X dynamic microphone and the M 90 PRO X true condenser microphone complete the new systems, targeting live streaming or recording, at home or in the studio.

The minimalistic M 70 PRO X and M 90 PRO X microphones are flexible designs, designed for a natural response, detailed nuances and an exceedingly low-noise signal at all volumes. An elastic system suspension and microphone spider will audibly help reduce unwanted sounds (e.g., footsteps, hand grips or keyboard noise). An additional pop filter is included to ensure that breathing noises and harsh plosive sounds are softened. A low-noise operational amplifier and a precise voltage regulator create a clear, warm sound, while the extended upper frequency range picks up every detail of the voice.

The new AKG Ara supports two pick-up patterns, and has a mic mute and headphone volume knob located on the front for easy access, benefiting from the included yoke and base stand.

beyerdynamic M 70 PRO X dynamic microphone and the M 90 PRO X true condenser microphone

The new beyerdynamic product series features a minimalist black design consisting of premium and interchangeable parts such as capsule, circuit board or connector. The high-quality aluminum housing and the robust steel mesh protect the microphone and reduce the weight of the PRO X microphones, allowing both models to be easily attached to any tripod or microphone arm.

Mobile Creation Microphones

The year was also rich in new products for mobile creation, even if many times those products also need to serve the production needs at home or in field production in multiple interior environments.

Sony Electronics announced the ECM-W2BT Multi-interface, shoe compatible wireless microphone and ECM-LV1 compact stereo lavalier microphone, creating a high-quality wireless audio recording solution to meet the needs of today's vloggers, YouTubers and video creators. The ECM-W2BT wireless microphone, using an advanced omnidirectional mic capsule, enables low-noise digital sound recording by connecting the receiver to a Sony camera equipped with a Multi Interface Shoe compatible digital audio interface. The ECM-W2BT also allows audio recording flexibility by increasing the stability of the wireless connection between microphone and receiver, even with obstacles or in crowded environments with additional wireless activity. In locations with good visibility, communication is possible at up to 200m, significantly increasing the flexibility of audio recording. The audio transmission uses the new Qualcomm aptX Low Latency codec.

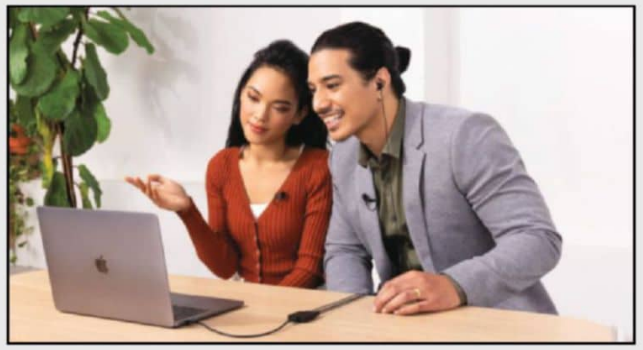
The ECM-W2BT mic supports stereo audio input and an external 3.5mm stereo mini-jack to allow for stereo sound pickup, so it can be used in combination with a stereo-capable lavalier microphone such as the ECM-LV1. The ECM-LV1 is an unobtrusive microphone that can easily attach to the subject's clothing and record sound with increased realism.

Fulfilling similar goals, but with a simpler and more limited-range concept, Mackie launched the OnyxGo Mic, a new clip-on lapel design that transmits audio wirelessly via Bluetooth to mobile devices, working in combination with a dedicated OnyxGo app.

The Mackie OnyxGO Mic is an integrated, compact, wireless mic and app, providing a completely self-contained, no-fuss wireless mic system, with no need for a receiver or added cables.

Sony surprised very positively with its ECM-W2BT Wireless Microphone and ECM-LV1 Compact Stereo Lavalier Microphone.

Building on the performance of existing designs, the new Audio-Technica lavalier mics, the BP898 and BP899, offer significant upgrades but remain affordably priced.



The Røde Microphones AI-Micro and the Røde Reporter app is a perfect combination for two-person interviews and mobile podcasting.

Users can simply pair it, clip it on, and they'll be ready to start recording or making calls. The companion OnyxGO app offers powerful capabilities like phone call recording, noise reduction, voice effects and EQ, AI-powered subtitle generation, and the interesting ability to mix the user's voice with Bluetooth audio.

For those users looking for a pro-level lavalier microphone for multiple applications, Audio-Technica introduced the low-profile BP898 Subminiature Cardioid Condenser and BP899 Subminiature Omnidirectional Condenser lavaliers. The two new lapel microphones are direct replacements and upgrades to the existing AT898 and AT899 models and are available in a number of different terminations (for wired or wireless use) that suit them to the entire range of lavalier mic applications. The BP898 is available in five black-finished models, all with permanently attached cables and varying termination and power configurations, while the BP899 is available in a total of 17 different models, all with permanently attached cables, varying termination and power configurations, with most available in two available sensitivities and black or theater beige finishes.

Both microphones have a low-profile design (a mere 5.3mm in diameter), which makes them ideal for on-camera work and

Rounding off the launch of its MKE 400 compact shotgun microphone for cameras and the XS Lav clip-on microphones for mobile phones and computers, Sennheiser has launched various Mobile Kits, which include a Manfrotto PIXI Mini Tripod and Sennheiser Smartphone Clamp.

any applications requiring minimum visibility. If up-close, directional miking is needed, the BP898 with cardioid capsule is recommended. The directionality of the BP898 cardioid element has a pronounced proximity effect when used at close range, enhancing the user's voice. If the mic needs to be used at a greater distance or a less focused pickup is desired, the BP899 with an omnidirectional capsule is recommended.

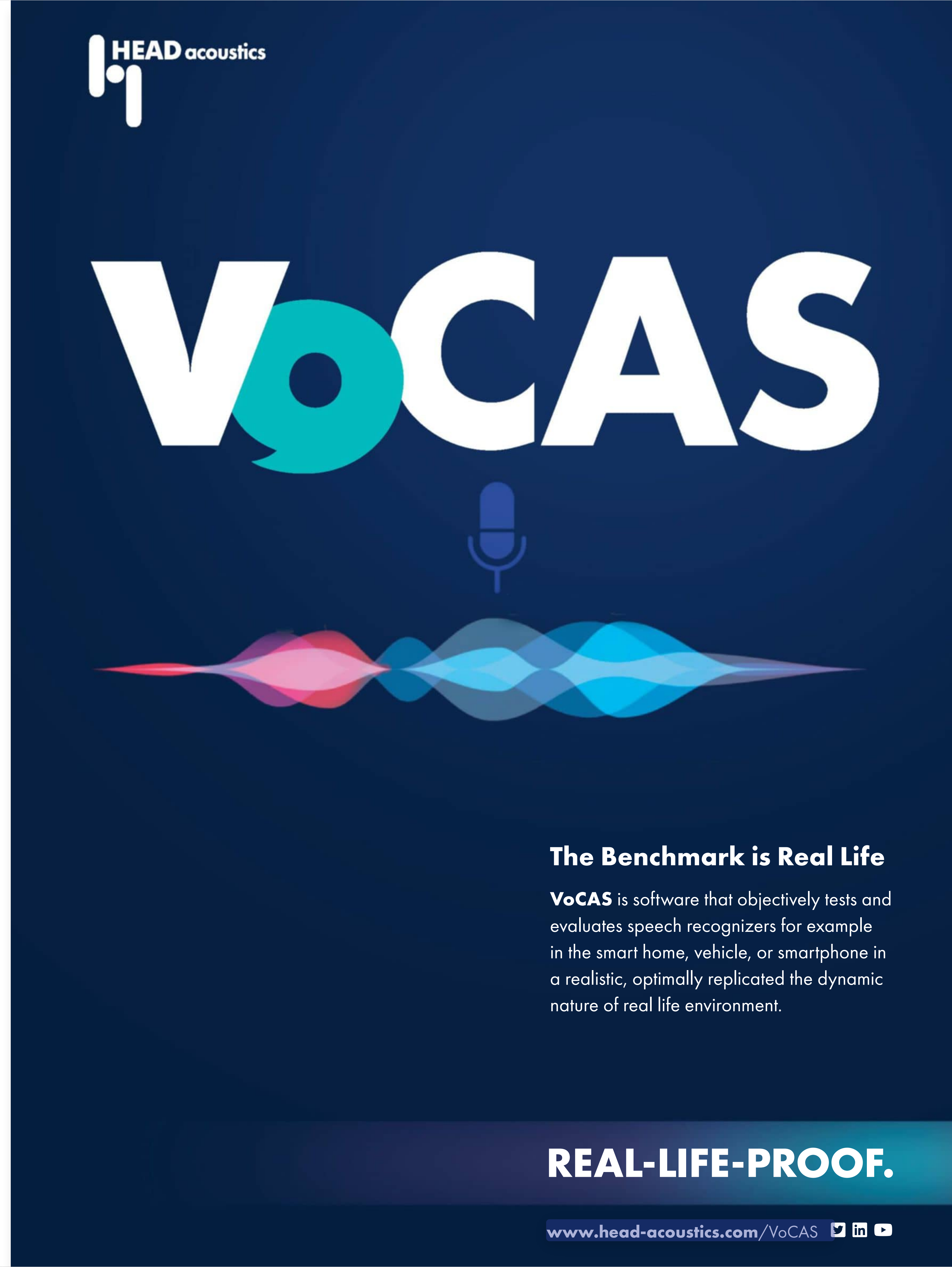
The BP899 features an open-diaphragm design for maximum frequency response with protective mesh that repels sweat and other moisture and lasting performance. It includes a resonance cap that can be applied to the mic for greater intelligibility and to direct sweat and moisture away. The BP899 is also available in low-sensitivity versions, for dynamic theater performances or other high-SPL applications.

And talking about lavalier microphones, the most innovative product in this area among those launched in 2021, has to be the AI-Micro, the new ultra-compact dual-channel interface from Røde Microphones. A great design for journalists, reporters, content creators, and anyone who needs to combine two microphones in a single recording device, the AI-Micro is another valuable product from the Australian brand, featuring two auto-sensing inputs for connecting virtually any microphone with a 3.5mm output, high-power headphone output for clear, zero-latency audio monitoring and playback, and a universal USB output for use with both iOS and Android devices.

Sennheiser also launched an interesting system for mobile recording. Following the introduction of the MKE 200, an affordable directional on-camera microphone, Sennheiser announced the MKE 400, a compact shotgun mic that delivers professional results, with a similar design to the MKE 200. Those directional microphones are complemented with the new XS Lav omnidirectional clip-on lavalier microphone and a 2.4GHz wireless transmission system. And to be able to offer a complete set of solutions for today's bloggers and creatives that capture video and audio with smartphones and DSLR cameras, Sennheiser teamed up with Italian brand Manfrotto to create a series of new Mobile Kits.

The new Sennheiser MKE 400 is a highly directional compact shotgun microphone that uses an acoustic interference tube, providing a super-cardioid polar pattern. The MKE 400's shock-mounted capsule minimizes handling noise, while the cleverly designed microphone housing doubles as a standard windscreen. For outdoor use, a furry windshield is included, and the microphone's low-cut filter also does its part in removing problematic low frequencies, such as the hum from air-conditioning systems or the rumble from wind.

Together with the MKE 400, Sennheiser introduced a new affordable omnidirectional lavalier microphone. The new Sennheiser XS Lav is powered by the smartphone or computer, which will automatically switch from the internal mics. It comes with a 2m cable using a standard 3.5mm jack or USB-C (ideal also for videoconferencing), together with a USB-C to USB-A adapter for legacy products. The package includes a microphone clip, removable foam windshield and a draw-string storage pouch. The XS Lav USB-C Mobile Kit additionally contains a Sennheiser Smartphone Clamp and Manfrotto PIXI Mini Tripod.



On the Edge

TinyML and Voice Recognition Technology Are Transforming Industries

By
By Vikrant Singh Tomar
(CTO, Fluent.ai, Inc.)

Consumers increasingly expect voice-enabled devices, such as home assistants or wearables, to operate offline and in a secure manner no matter the environment. Rising demand for portable, light footprint voice recognition devices is fueling a growth of the tiny machine learning (TinyML) field. This article explains the impact for embedded voice and the overall audio industry.

Technology is only useable if it can be deployed where users are, and advancements in tiny machine learning (TinyML) and voice recognition are bringing it to the edge.

Synonymous with edge AI, TinyML has been described by Harvard researcher Matthew Stewart, PhD., as "the intersection of machine learning and the embedded internet of things (IoT) devices." The field of TinyML grew from the concept of IoT, which traditionally operates under the assumption that data will be sent from devices to the cloud for processing. This method now raises concerns across enterprises about data security, latency and storage, energy efficiency, and connection dependency. In contrast, devices on the edge process data locally, taking the cloud and data centers out of the equation (**Figure 1**).

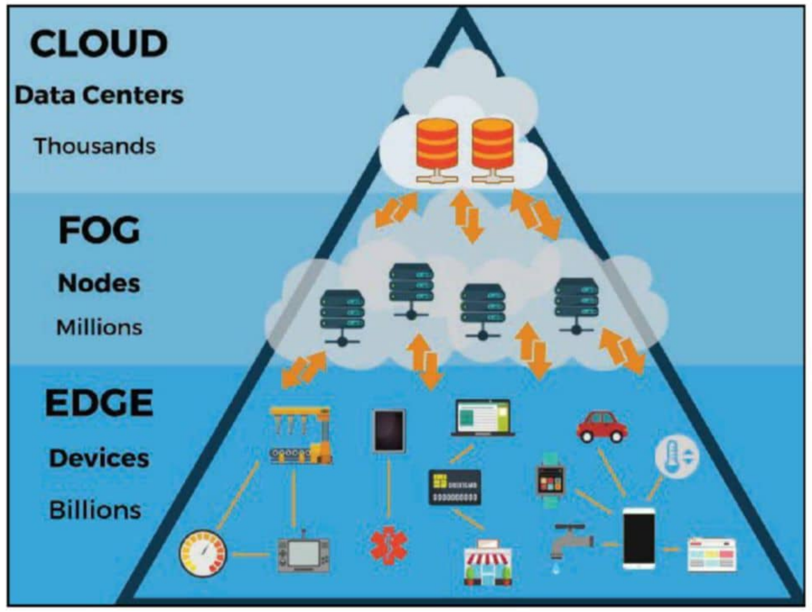


Figure 1: Hierarchy of the cloud (Image courtesy of eBizSolutions)

At the forefront of this evolution, usage of TinyML technologies and applications, specifically those utilizing voice, is growing as demand for specialized edge AI devices across industries and consumer spaces rises.

Evidence of this increased focus on low-latency, power-efficient edge AI technology capable of addressing the concerns caused by dependency on the cloud is ABI Research's expectation that a total of 2.5 billion devices will be shipped with a TinyML chipset by 2030.

How TinyML Solves the Pain Points of Current ML Technology

Machine learning (ML) was defined in the 1950s by AI pioneer Arthur Samuel as "the field of study that gives computers the ability to learn without explicitly being programmed." As modern machines are expected to handle more complex tasks ranging from language translation to powering autonomous vehicles and beyond, data storage and programming becomes more complex.

ML begins with data and from there programmers select a machine learning model to use, supply the data, and allow the computer to find patterns or make predictions. The more data a machine is able to train off of, the better the program, and programmers can change model parameters over time to push the machine to generate more accurate results.

Traditional ML technology has been trained to process large volumes of information to complete a variety of complex tasks and has a dependency on servers or the cloud for mass data storage, raising issues in enterprise and consumer spaces ranging from security to latency and energy efficiency.

Privacy concerns are front and center for connected devices (e.g., smart home appliances and voice assistants) that are always listening for a wake word in a user's home. Concerns regarding privacy and compliance are enhanced when such devices are used in enterprise settings and secure information may be transmitted. On the user side, even small amounts of lag can alter an experience when interacting with smart TV remotes or wearable devices (e.g., fitness trackers or headphones). In an industry setting, high latency negatively affects efficiency. Energy efficiency is also key in AI-enabled devices to avoid the need for constant recharging or docking as most consumer devices are battery-dependent.

TinyML and Edge AI Technology Offer a Solution to Cloud-Dependent Devices

Low Latency/Offline Execution: Devices on the edge work offline and provide end users the convenience of operating their devices without an Internet connection. Cloud-free models ensure devices can run offline in (almost) real-time as there is no need for a round-trip latency that could be slowed by network connectivity levels.

Data Security: Data processed locally on the device does need to be sent to the cloud for processing or stored on servers. As concerns grow over privacy and smart devices are increasingly deployed in sectors that handle confidential information such as hospitals, offices, or factories, this factor becomes more relevant.

Energy Efficiency: Processing real-time data on the edge not only protects privacy and enables low latency, but also saves power. Edge AI is designed to have a low footprint and work on battery-powered devices for years at a time.

Despite recent developments in ML, finding an optimal solution for a given task remains challenging and time-consuming. It often requires significant effort in designing and tuning the neural architectures, particularly for embedded TinyML solutions, which need specific models due to limited computational resources.

One of the newest innovations in TinyML addressing this issue, Guided Neural Architecture Search with Imitation Learning (GNASIL), uses a ML tool called a soft-actor reinforcement learning (RL) agent to discover new neural networks quickly and effectively. It combines deliberate planning with predictions of the RL agent and chooses the best action at a time. In GNASIL, the reward function combines the accuracy of the newly discovered neural network structure on a validation set and the computational footprint of the designed architecture in floating-point operations per second (FLOPS).

Experiments on a series of on-device speech recognition tasks demonstrated GNASIL is performing as intended (**Figure 2**). The innovation can design neural models with competitive performance both in terms of discovery speed and

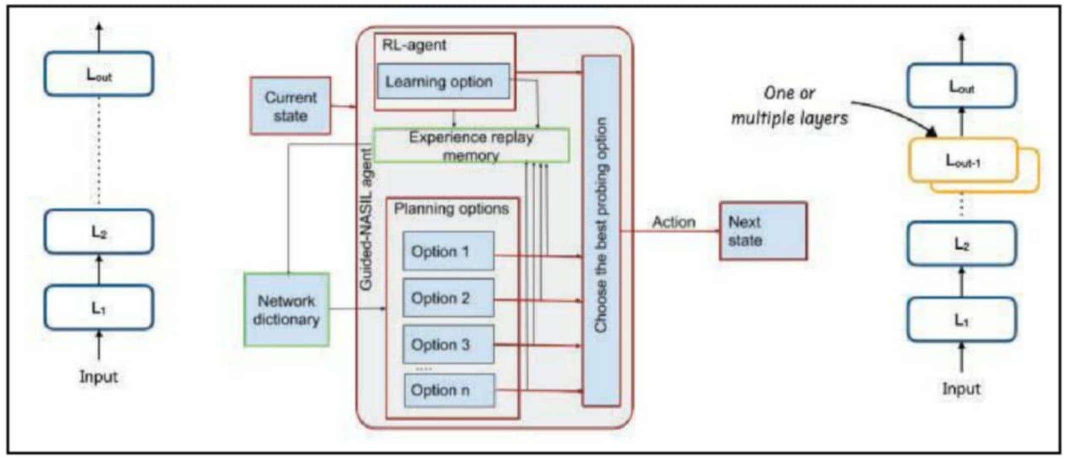


Figure 2: Outline of the planning and learning components of GNASIL

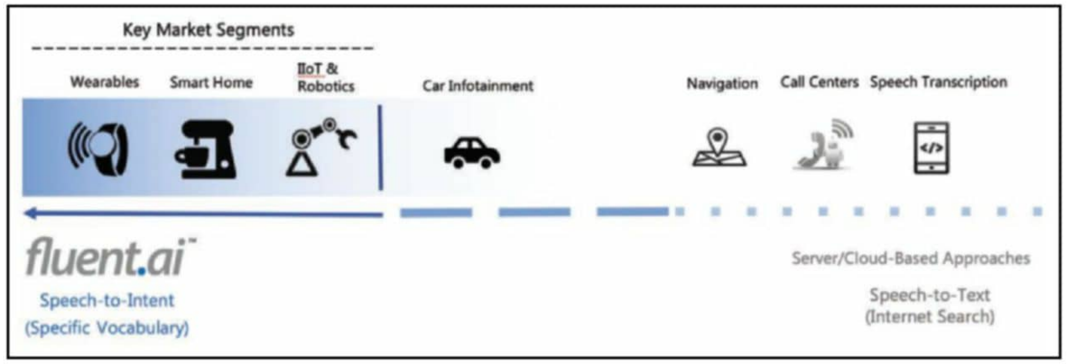


Figure 3: Within five years, voice recognition technology on the edge is going to permeate everyday life.

the accuracy of the discovered architectures, all within the predefined (FLOPS) restrictions, helping design and optimize neural network architectures more quickly than before.

Deploying TinyML for the Enterprise

At the intersection of ML and IoT, TinyML is actively transforming enterprises as we know them.

McKinsey researchers predict IoT will have a potential total economic impact of as much as $11 trillion by 2025, and Gartner expects the number of worldwide IoT-connected devices to increase almost threefold from 2018 to 43 billion devices by 2023. Combine these projections with the aforementioned expectations of ABI Research and the effect is a sector of the industry that "extends beyond the borders" of traditional technology.

The growth of TinyML has resulted in an increased interest in deploying speech and voice recognition devices across industries. TinyML hardware, algorithms, and software capable of performing on-device data analytics enable a variety of always-on use cases, targeting battery-operated devices in an array of industries from manufacturing to healthcare.

European manufacturer BSH deployed Fluent.ai's offline, low-latency speech-to-intent AI technology in one of its German factories, estimating up to 75% to 100% efficiency gains on the assembly line due to voice automation.

Conventionally, factory operators must press physical buttons at their workstations to move appliances along the assembly line. To streamline efficiency and improve ergonomics, Fluent.ai developed a voice-control solution for workers to operate heavy machinery by speaking into a headset that is connected to an embedded voice recognition solution. Initial implementation of the solution cut down the transition time between stations by more than 50%.

Designed with loud factory settings in mind, the technology is robust and has the ability to cancel out background noise. It is also multi-lingual and accent-agnostic, allowing the technology to be deployed anywhere, regardless of location or spoken language.

In healthcare, Edge Impulse and Arduino applied an optimized TinyML model to build a cough detection system that runs in real-time in under 20kB of random access memory (RAM) in response to the COVID-19 pandemic. Further TinyML concepts have been designed that could analyze a patient's heart rate, body temperature, and respiratory rate to predict a patient's health condition.

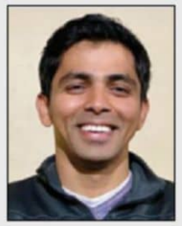
Hardware partners such as Dell are also supporting TinyML with specialized hardware that hosts and integrates intelligent data at the edge, bringing innovation to IT and business stakeholders. It not only assists in delivering an intelligent insights platform for organizations, but also supports 5G and WiFi 6 innovations.

TinyML and the Future of Connected Devices

Achieving high efficiency on low power is the ultimate benefit of TinyML devices and innovations, such as GNASIL, are laying the groundwork for more optimized models. Within five years, voice recognition technology on the edge is going to permeate everyday life from smart appliances to wearables and office meeting rooms to restaurants as speed, convenience, efficiency, and privacy are prioritized (**Figure 3**).

The industry is already seeing more specialized chips being designed, and tiny machine

About the Author



Vikrant Singh Tomar is Founder and Chief Technology Officer of Fluent.ai, Inc. He is a scientist and executive with more than 10 years of experience in speech recognition and machine/deep learning. He obtained his PhD in automatic speech recognition at McGill University, Canada, where he worked on manifold learning and deep learning approaches for acoustic modeling. In the past, he has also worked at Nuance Communications, Inc. and Vestec, Inc. as a Research Scientist.

learning algorithms and low-power hardware (microcontrollers) have evolved to the point that it's possible to run sophisticated models on embedded devices and speech technology is becoming the rule, not an exception.

Privacy and cybersecurity are two of the biggest concerns emerging from the surge of the voice recognition market and interconnectivity to the cloud, and TinyML devices are being designed to meet compliance obligations in consumer and enterprise settings. In the near future, specialized devices will be able to hear and alert when machines are malfunctioning or even when a faucet is leaking—all in an offline capacity.

Traditional speech understanding solutions operate in two distinct steps:

- Speech is transcribed into text in the cloud.
- Natural language processing is applied to the text to determine the user's intent.

Cutting-edge voice recognition technology directly maps a user's speech to their desired action without the need for speech-to-text-transcription or a cloud connection. A text independent approach enables the development of speech recognition systems for any language or accent that run completely offline and work accurately in noisy environments.

At the forefront of speech-to-intent technology, Fluent.ai employs unique deep neural network algorithms to map speech. The technology learns by directly associating semantic representations of a speaker's intended actions with the spoken utterances (**Figure 4**).

The current AI behind the technology is being trained to better understand accents from conception, and challenges from background noise are being mitigated to build more robust speech recognition experiences. The models are based on the concept of vocabulary and language acquisition in humans.

A text-independent approach enables the development of speech understanding models that can learn to recognize a new language from a small amount of data compared to cloud-based systems and enables end users to interact with their devices in a more natural and intuitive way.

Embedded voice solutions support an end user's freedom to travel anywhere and do anything while protecting privacy and keeping their device's smart voice capabilities with them wherever they go. From smartwatches and fitness trackers to headphones and earbuds, voice technology on the edge gives users the flexibility and hands-free convenience of accurate voice command control in any environment. Noise robust solutions also allow for usage in the noisiest locations, whether it be a busy street or a family gathering.

Highly portable and light footprint, offline speech recognition solutions are the technology driving today's TinyML voice recognition demand surge. Systems capable of running on microcontrollers and digital signal processors (DSPs) will lead the evolution of voice recognition on small devices.

Voice has become a mainstream tool for enterprises, apps, and digital platforms due to its ability to simplify the user interface. As processes continue to be streamlined because of advancements in TinyML and voice recognition solutions, offline devices capable of executing specialized commands are going to permeate every industry.

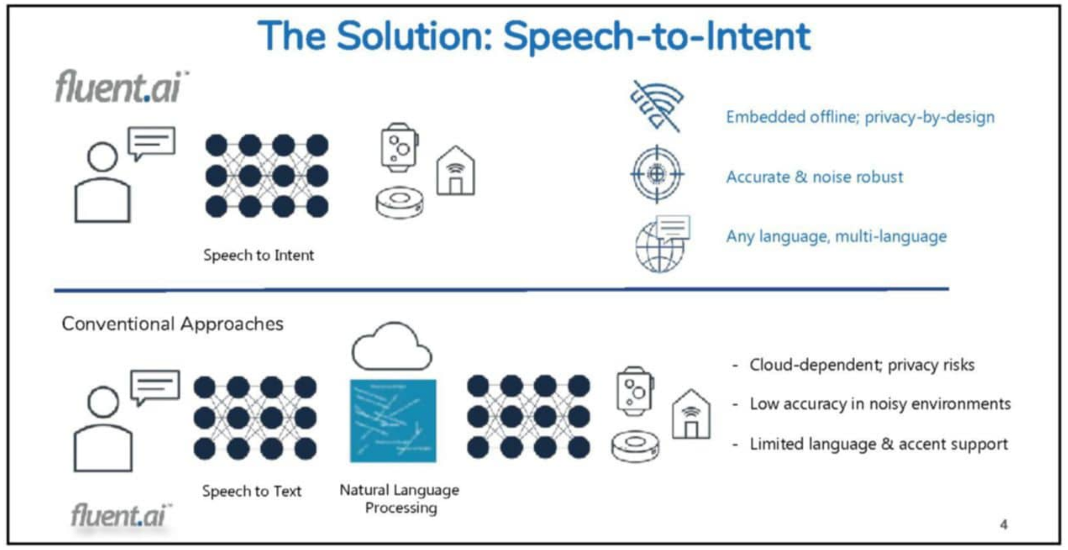


Figure 4: Comparison of Fluent.ai's TinyML speech-to-intent technology and a conventional cloud-based speech-to-text approach

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Quantification of Speech Intelligibility

In this article, our columnist discusses the history of speech intelligibility rating systems and the measurement techniques that evolved into today's international standards.

By
Richard Honeycutt

Photo 1: Enlarged mouths of Greek theater masks acted as megaphones.

No one is sure just when people began gathering together for social events or meetings. Likely groups of hunters assembled to plan a hunt, and groups of families gathered after the hunt to enjoy the results of the hunters' success. But the earliest assemblies involving speech programs that we know anything about were religious meetings at which stories would be told or scriptures would be read, and early theater presentations.

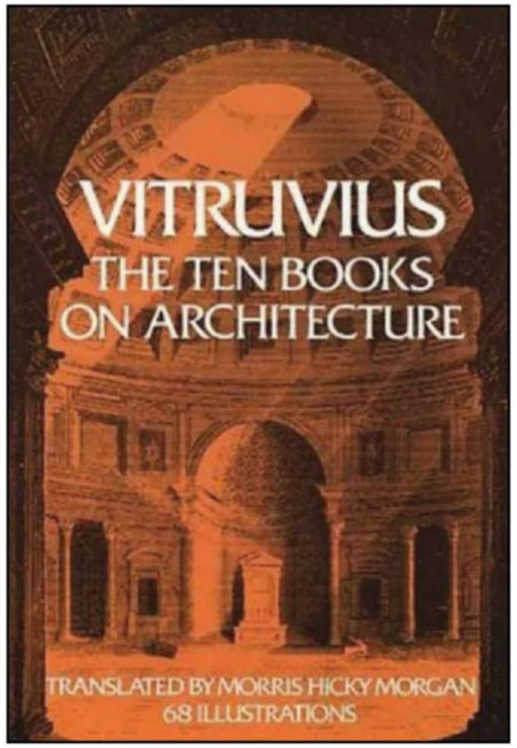


Photo 2: Roman architect Marcus Vitruvius Pollo published books on architecture in the First Century CE.

Ancient Greeks performed plays on raised platforms (e.g., threshing floors) with audiences seated around on surrounding elevated ground. In order to facilitate the audience's ability to understand the words spoken by the actors, masks with grossly enlarged mouths were used (**Photo 1**). These acted as megaphones, matching the high acoustical impedance of the human larynx, throat, and mouth to the lower acoustical impedance of the air, thus increasing acoustical radiation resistance and consequently, efficiency and level of the projected voice [1].

Architects learned other means for increasing speech level for audiences as drama, lectures, and religious preaching became more prominent parts of daily life. However, audiences were not always able to understand speech even if the sound level was sufficient. In book 5 of his *Ten Books on Architecture* (**Photo 2**), first-century architect Marcus Vitruvius Pollo discussed concepts that early English translations rendered as "circumsonance" and "ressonance," which today we would call "reverberation" and "echoes." These were recognized as being detrimental to speech intelligibility.

During the 19th century, efforts toward acoustic enhancement of auditoria and concert halls were principally directed at providing reflecting surfaces (sounding boards) to concentrate the energy from a speaker, singer, or instrument(s) toward the audience. In 1853, Dr. J. B. Upham of Boston, MA, undertook a careful study of the acoustics of the Boston Music Hall, culminating in an essay on reverberation and resonance.

American physicist Joseph Henry presented papers to the American Association for the Advancement of Science in 1854 and 1856, and a new lecture room in the Smithsonian Institution was built according to conclusions of his experiments and theory. The results were said to be highly satisfactory.

Architect T. Roger Smith published his book *Acoustics of Public Buildings* in 1861, in which he discussed the differing acoustical requirements of speech and music. John Tyndall and John William Strutt (Lord Rayleigh) investigated the control of reverberation in rooms. But the definitive work that is universally agreed to mark the genesis of scientific acoustics was that of Harvard University Physics Professor Wallace Clement Sabine.

In the very last years of the 19th century, Sabine was tasked with remedying the horrific acoustics of the Fogg Lecture Hall. He began with a quantitative study of the nature of reverberation and the physical factors controlling it. His equation for predicting the reverberation time of a room is still widely used.

Aside from Sabine's seminal work on the prediction and measurement of reverberation, his recognition of its importance in speech intelligibility, while not pioneered by him, certainly became a bedrock of modern acoustical engineering's work in understanding intelligibility.

Speech Intelligibility Rating Systems—AI

After Dr. Harvey Fletcher began work at Western Electric in 1916, he—along with researchers—started the process of defining the measurement standards for loudness and articulation [2]. Fletcher's work on articulation, which is related to speech intelligibility, focused mainly on the signal-to-noise ratio (SNR) at the listener's ear. Thus, SNR was recognized along with sound level and reverberation as important in affecting speech intelligibility. Since Western Electric was primarily interested in improving telephone communication, the acoustical performance of concert and lecture halls were not part of Fletcher's work. Neither were the effects of bandwidth and distortion.

The 1947 publication of his 1921 work on an Articulation Index (AI)—the percent of speech sounds interpreted correctly during a telephone conversation—can be viewed on the *Journal of the Acoustical Society of America* website (see Resources). The AI is calculated from measured 1/3-octave or octave-band SNRs, which are then weighted according to their effect upon intelligibility.

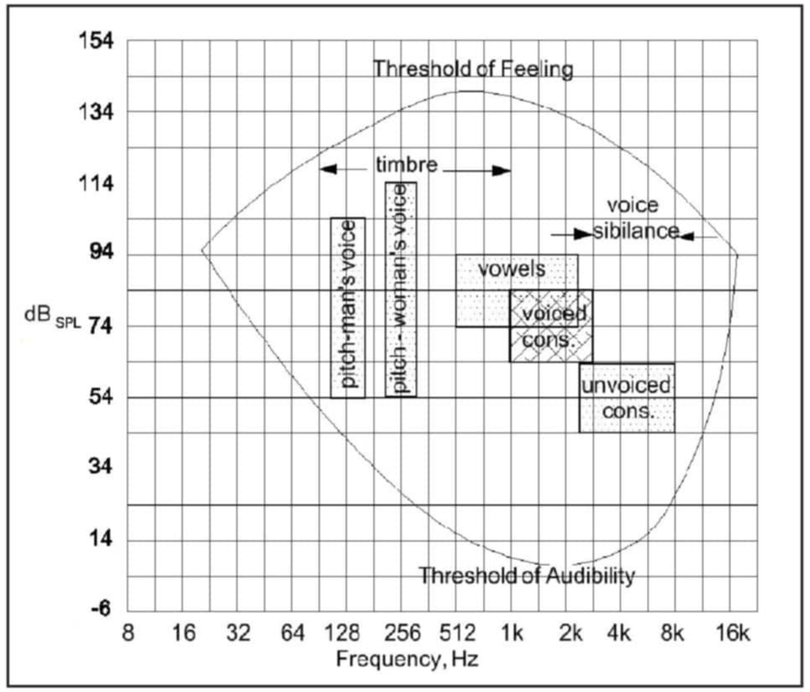


Figure 1: Vocal timbre, speech vowels, consonants, and sibilants are associated with different frequencies in the audible spectrum.

The resulting number is scaled from zero (no intelligibility) to unity (perfect intelligibility).

More general testing of speech intelligibility involved trained talkers pronouncing words from a standard list, with listeners writing down what they heard. The percentage of correctly understood words provided quantification of speech intelligibility. This method could be used for acoustical as well as electroacoustical testing, but was time-consuming and expensive. It also suffered from some ambiguity, since if the sample speech consisted of isolated words, the scores tended to be lower than if sentences were used, as listeners could use context to help identify sounds that were not completely clear.

Speech Intelligibility Rating Systems—%ALcons

In 1971, Victor M. A. Peutz published "Articulation Loss of Consonants as a Criterion for Speech Transmission in a Room" (see Resources),

About the Author

Dr. Richard Honeycutt fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject when Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical instruments and sound systems. He has been an active acoustics consultant since he received his PhD in electroacoustics from the Union Institute in 2004. Richard's work includes architectural acoustics, sound system design, and community noise analysis.

in which he introduced a reverberation-based articulation index. The use of consonants is based upon the easily observed fact that consonants have more effect upon speech intelligibility than do vowels (**Figure 1**). Peutz's paper introduced an "architectural equation" for predicting the articulation loss of consonants (%Alcons):

$$\%ALcons = \frac{200 T_R D_2 (n+1)}{VQ}$$

where T_R is room reverberation time, V is volume of room, D_2 is distance between loudspeaker and farthest listener, Q is the directivity of the loudspeaker, and n+1 accounts for the effects of multiple loudspeakers adding to the reverberant field.

So far, we've discussed the effects of audio level and reverberation upon speech intelligibility. Background noise is another important factor. In fact, one school of thought considers reverberation to be a form of noise, and bases assessments of speech intelligibility upon SNR. Actually, a variation of the architectural %Alcons equation incorporating noise was developed, but since it gave different predictions from the original equation, it is not commonly used. As you can infer from Figure 1, the effects of peaks and dips as well as low-frequency and high-frequency limitations in sound system frequency response can also impact speech intelligibility.

Speech Intelligibility Rating Systems—STI

In an effort to include all detrimental effects upon speech intelligibility, Tammo Houtgast and H.J.M. Steeneken of the government's TNO research laboratories in Netherlands published a paper on a very different approach to quantifying speech intelligibility: the Speech transmission Index (STI) [3]. From the abstract of their paper in *Acta Acustica*, the method is described thus:

"When a signal is passed through an enclosure, the envelope of the received signal is in general a smoothed version of the original envelope. Traditionally, this effect of the enclosure is quantified by its reverberation curve. We suggest that the relation between the input and output envelopes can also be described by the action of a filter characteristic which is effective upon the input envelope. In the case of smoothing, the filter is essentially a low-pass filter. This filter characteristic (Modulation Transfer Function, MTF) is a specific feature of a given enclosure, applicable to the envelope of any input signal.

We investigated the relevance of this MTF for quantifying speech intelligibility in enclosures. For a great number of transmission paths, set up in the laboratory (reverberation and single echoes, combined with several degrees of noise interference), we determined (1) the MTF for modulation frequencies from 0.25Hz to 25Hz, and (2) the intelligibility score from listening tests (PB-word score) [PB stands for "phonetically balanced"].

It appeared that, for each condition, a single value could be deduced from the MTF, which correlates highly with the PB-word score. The data suggest that the MTF may serve as a convenient predictor of intelligibility in room acoustics, with an accuracy of about 4% in PB-word score."

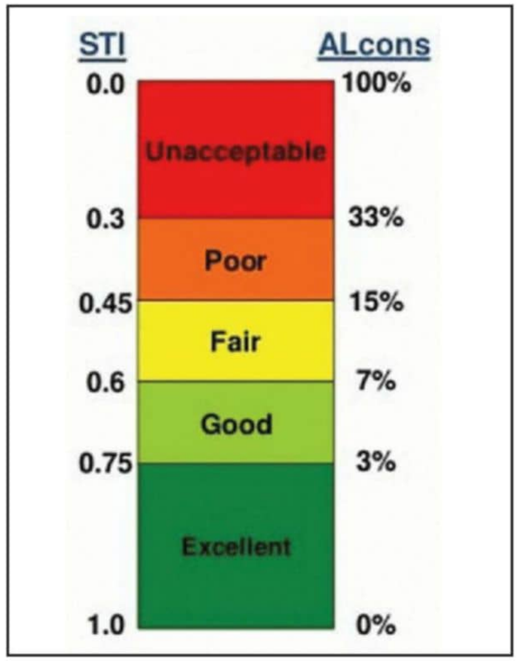


Figure 2: This chart shows the approximate %ALcons and STI ratings that correspond to different levels of speech intelligibility.

Speech energy in the bands from 125Hz to 8kHz are effectively modulated over a range of frequencies from 0.63Hz to 12.5Hz. The "smoothing" mentioned in the paper's abstract is correlated with a reduction in speech intelligibility. Each octave band contributes to intelligibility to a specific extent, so by weighting the MTF values of the various octave

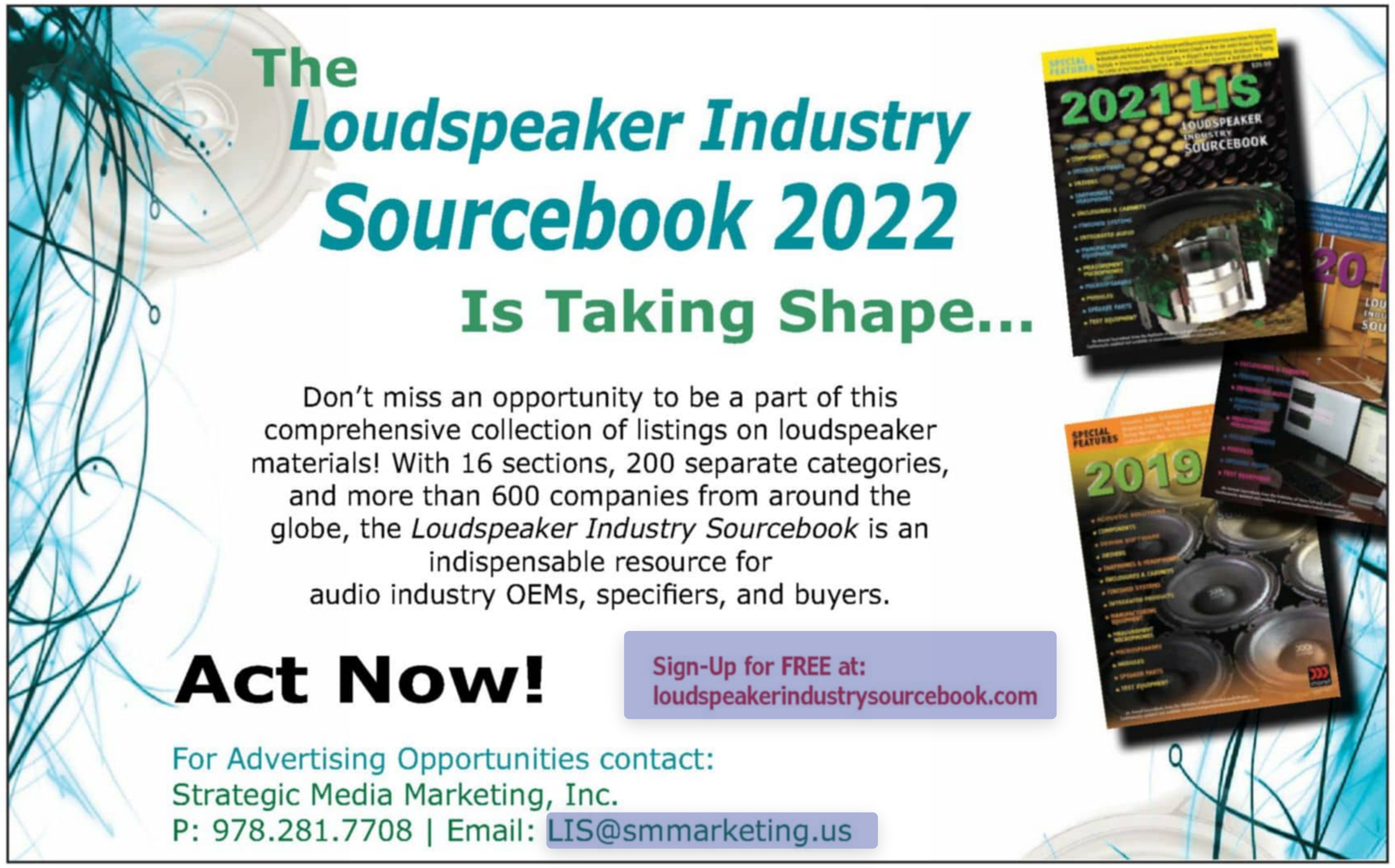
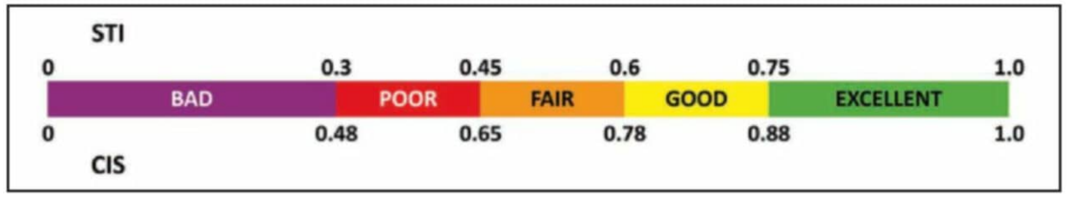


Figure 3: This chart compares the STI and CIS scales.



bands by each band's contribution to intelligibility, the STI is computed.

Since %Alcons and STI are based on very different analyses, it is not strictly possible to convert one rating to the other. However, **Figure 2** shows a chart that shows the equivalent ratings from the two systems, based upon their perceptual effects. Most computerized acoustical modeling systems can predict both %Alcons and STI.

Photo 3: The Bedrock SM50 measures full STI (65-second measurement) as well as quick STIPA for screening, and functions as a sound level meter and RTA.

References

[1] R. A. Honeycutt, *Acoustics in Performance*, Elektor International Media, B. V.; London 2018.

[2] AT&T internal documents, quoted by Jont B. Allen, *Journal of the Acoustical Society of America*, Volume 99, No. 4, Part 1, pp. 4-96

[3] T. Houtgast and H.J.M. Steeneken, "The Modulation Transfer Function in Room Acoustics as a Predictor of Speech Intelligibility," *Acta Acustica 28*, pp. 66–73, 1973.

Resources

"A Modern Version of the Old Theory of Telephone Quality," *The Journal of the Acoustical Society of America*, Volume 19, Issue 1, June 2005, https://doi.org/10.1121/1.1902525

V.M.A. Peutz, "Articulation Loss of Consonants as a Criterion for Speech Transmission in a Room," Audio Engineering Society, Volume 19, Issue 11, December, 1971, www.aes.org/e-lib/browse.cfm?elib=1820

T. Roger Smith, *Acoustics of Public Buildings*, 1861.

Other factors besides level, noise, and RT affect speech intelligibility. There are person-specific factors including, talker enunciation and rate of speech and listener acuity. These must be taken into account in the design of public venues. For example, lecture rooms in retirement homes should be designed for an STI of at least 0.65-0.70, preferably 0.75 or higher. Inexperienced talkers should be instructed on how to speak slowly and clearly, and how to properly use a microphone.

Additional acoustical parameters affecting speech intelligibility include direct-to-reverberant (D/R) ratio and echo. D/R ratio is specified by the clarity (C35, C50, etc.) or definition (typically D50). These measurements take beneficial early reflections into account. None of the measurements considers the documented speech intelligibility decrease caused by excess sound levels: above 90dB, the ear produces intermodulation distortion (IMD) that degrades intelligibility.

Echoes—sound delayed by more than 50ms to 80ms—reduce intelligibility, as anyone who has tried to understand the announcer at a pre-1970s stadium can attest.

Measurement of Speech Intelligibility

Many instruments provide the capability to measure the speech intelligibility of a venue, and with the adoption of international standards—International Electrotechnical Commission (IEC) Standard 60268 and ANSI/ASA S3.2-2009—instruments from different manufacturers have become more consistent. However, measurement of %Alcons is usually done using only the 2kHz octave band, leading to significant errors in measurement results. Measuring and calculating the full STI is processor- and time-intensive. Consequently, the Rapid Speech Transmission Index was introduced during the 1980s. Using only the 500Hz and 2kHz octave bands, RaSTI measurements were found to be inaccurate under some conditions, especially for inexpensive sound systems.

Recognizing the limitations of different intelligibility rating scales, engineers in Europe devised the Combined Intelligibility Scale (CIS), which was widely adopted in Europe, although RaSTI measurements still appeared in some engineering and licensing specifications. **Figure 3** illustrates the values of STI and CIS that correspond to various perceptual intelligibility ratings.

In late 2001, a new measurement protocol—STIPA (STI for PA systems) was introduced. Some STIPA instruments display both the displays both CIS and STI readings. The STIPA protocol requires two modulation samples per frequency band, as compared to 14 for a full STI measurement. This explains why STIPA measurements can be made more rapidly than STI measurements, but it also leads to somewhat reduced repeatability and accuracy. **Photo 3** and **Photo 4** show two multifunction audio measuring instruments that include STI measurement capabilities.

Making Good Measurements of SI

In practice, no matter what rating system is used, the speech intelligibility of a room should be measured using natural acoustics (sound system powered off) before measuring with the sound system operating. Comparison of the "natural" and "amplified" result will allow you to quantify the effect of the sound system upon speech intelligibility: Not all sound systems help intelligibility!

Photo 4: The NTi XL2 functions as a STIPA meter, sound level meter, RTA, spectrum analyzer, and vibration analyzer.

Haptic Devices for Augmentation, Entertainment, and Wellness

By
Mike Klasco
(Menlo Scientific, Ltd.)

The music and the auditory world encompass not only airborne sound energy but also structure-borne energy, which is perceived by our haptic senses. The ability to interact with the world through vibration and touch is central to all of our lives, and now increasingly so. In experiencing live sound, our haptic perception contributes to the realism, and conversely haptics can strengthen the illusion and simulation of reality.

Sound is energy made by vibrations and at a concert you see the performers, hear the music, and feel the bass haptically through structure-borne and soft tissue sensations. While audio magazines focus on what we hear, and audio-video publications also detail what we see, haptic sensations are not a topic commonly "touched" by publications and yet they augment our senses.

In the January 2021 issue of *audioXpress,* we provided a thorough exploration of bass shakers and haptic devices with a focus on music and home theater reproduction. We promised to return to the topic and further explore tactile technology and the wide range of unconventional haptic applications.

The ability to perceive and interact with the world through vibration and touch contributes to the realism, and conversely haptics strengthen the illusion and simulation of reality. The smoothing vibrations of haptic therapies have proven in many clinical studies to calm the nervous system and improve heart rate variability (HRV) as a safe, non-invasive stress relief tool without drugs or side effects. Clinical studies have shown positive interaction with depression, PTSD, bipolar, insomnia, and autistic users.

Reproduction of this tactile component can be achieved through the use of a transducer popularly known as a "bass shaker" coupled, sized, and tuned into the specific platform. These transducers pass low-frequency vibrations into them so that they can be felt.

Bass shakers have the power to intensify everything while improving the sensory interaction. For gaming, home theaters, movie theaters, discotheques, and theme park rides, this "tactile" component is able to add a fourth dimension to reproduce the tactile impact of explosions, engine sounds, thunder, and other bone-rattling low-frequency content.

First popularized by Pioneer more than 40 years ago with its Bodysonic bass shaker, used everywhere from discos to car seats, today's haptics will give you a nudge when your smartphone is silenced, your Advanced Driver Assistance System (ADAS) will give you a poke when you are driving and feeling sleepy, or you might get a tactile "handshake" feedback from a touchscreen confirming your selection.

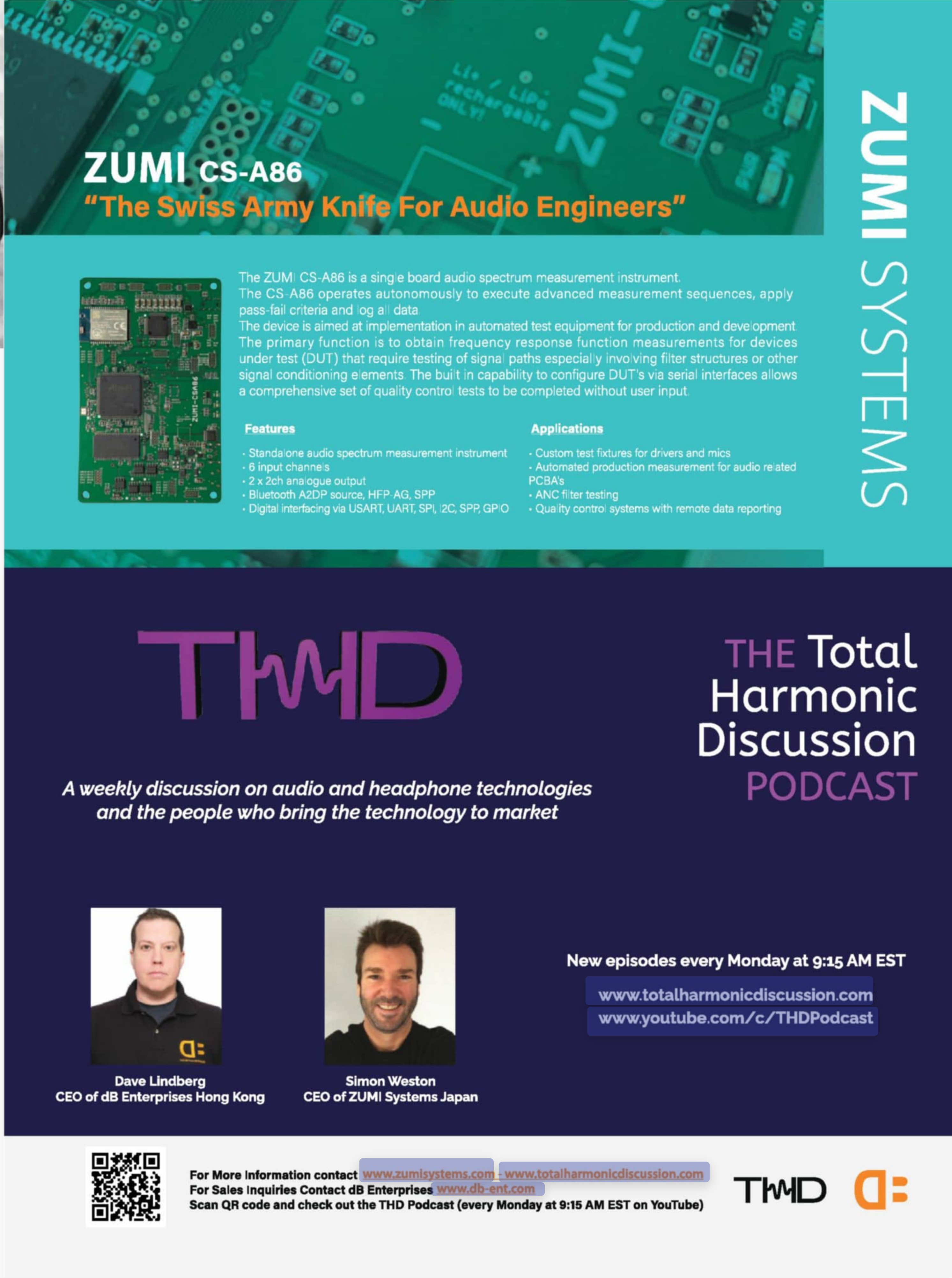
The functions and locations are truly all over the place with end users increasingly coming into contact with haptics in medical devices, smartphones/tablets, and other touchscreens for gaming, sport fitness, and more.

Haptic motors are what most of us initially encountered in flip phones using a micromotor with an off-center weight on the shaft. These encompass two common motor types: Eccentric Rotating Mass (ERM) vibration motors and Linear Resonant Actuators (LRA) vibration motors (also known as linear vibrators).

While the ERM haptic actuators are adequate to discreetly poke you in your pocket when getting a call during a meeting, these traditional haptic drivers only deliver simple vibration buzzes due to the narrow frequency range and non-agile response. For gaming and other audio applications, the latest efforts of these motor actuators are somewhat better, with effects ranging from subtle to sharp, and textured effects that simulate different surfaces and sensations.

The standardization of haptics for touchscreen user-interfaces is expanding in multiple consumer-facing products, from automobile dashboards, to smart appliances, and advanced game controllers, all incorporating "high-def" motor haptics.

Automotive applications for vibration transducers are omniscient. From haptic "shakers" built into driver's seats to ADAS driving awareness alerts, the automotive applications for haptics and vibration transducers are a multi-part article in itself. The next generation of automotive system engineering for the car interior expands to solutions for individual seats, the driver's touchscreen controls, which bump back (bilateral interaction) when you input is acknowledged so you do not need to visually confirm, and infotainment systems enhanced with sound and haptics in the headrest and the individual seats. There is no way to achieve this personalization and separation of programs and functions without most of the bass being transmitted haptically from the seats, all the while getting accident avoidance



bleeps from the general direction of the threat—not to mention the vibration cancelation from the active engine mounts.

Tactile Sensation for Calmness?

While the traditional application of haptics has been to turn up excitement, it has also been successfully applied to calming and wellness product categories as well. With more "excitement" now in our lives than anyone deserves, throughout the world people are turning to meditation, religion, therapy, massage, not to mention drugs and alcohol. We are in the Golden Age of stress, and haptics is a unique core technology for calming devices.

Pandemic confinements and quarantines are having a toll on many people's activities, routines, and livelihoods. Today there are significantly higher levels of loneliness and depression, and Covid-19 has increased stress levels throughout the world.

Haptics can serve as alternative or adjunct to drug therapies for persistent mental distress, such as depression, anxiety, obsessive-compulsive disorders (OCD), insomnia, depression, and high blood pressure. Touch is an integral part of the human condition. Maybe the brain's hardwired response to social touch can also extend into the realm of digital haptics. Petting your dog or stroking your cat makes you feel better (and the pet also). A mother's soothing touch comforts a child and a massage can help you let go of the day's stresses. This "feeling" is your Haptic perception and is divided into two different classes: Tactile and Kinesthetic. The texture you feel in your fingers, feet, and so forth are sensors on the surface tissues embedded in the skin and right underneath it, enabling your brain to feel vibration, pressure, touch, and texture.

The things you feel from the sensors in your muscles, joints, and tendons are classified as Kinesthetic. Tactile perception of live sound or even recorded music with powerful subwoofers or bass shakers actually shows up first just before the airborne sound component as the speed of sound through materials is faster than it is through the air.

With this background information, let's start our exploration of a half-dozen over-the-counter devices for wellness, grounding, and just chilling out and see how they explore haptic technology. Most of these "get out of a funky mood" use tactile transducers, but we will also check out electro-muscle-simulation and cranial electrotherapy stimulation gizmos.

Wellness tactile products are typically compatible with audio (audio/video) program sources as well as offered with specific program material. The purpose of haptic wellness devices is to induce (entrain the user's brain) into an altered state of consciousness, through generation of sensory, perceptual, and emotional experiences. Yet aside from calming, the various haptic devices can also induce alertness or just get you up and out of bed in the morning.

Binaural Beats and Brain Frequency Patterns

To experience a taste of calming simulation, you might search for "binaural beats" on YouTube or any of the music streaming services, and play these programs in the background while you read this article. Ideally experienced with a headphone but stereo speakers will still provide a sense of a sound-induced meditative state. Binaural beats are an emerging sound therapy making use of the right and left ear, each receiving a slightly different tone, yet the brain perceiving a single difference tone. The binaural tones should be at frequencies lower than 1,000Hz for the brain to detect the beat. The perceived tone is the frequency difference between the waves entering the left and right ear. For example, if the left ear registers a tone at 200Hz and the right ear registers one at 210Hz, the binaural beat is 10Hz—the difference between the two frequencies. You might also remember this is a phenomenon used on the organ foot pedals. The YouTube channels of binaural beats are usually accompanied by soothing kaleidoscopic transitioning eye patterns—all very new age. The purpose of binaural beats therapy may differ among individuals as some people may need help decreasing their anxiety, while others want to increase concentration or deepen meditation level.

There are many studies where one sense leaks to other perceptions. In one famous study, during a meeting, the interviewer was given a warm cup of coffee and in other cases a cold cup of coffee. When asked about the person being interviewed, the drink temperature had a significant impact in how warm or cold the person being interviewed was perceived by the interviewer.

As another example, we all tap our feet to a catchy beat, so it is plausible that our brain will pick up on a vibration and sync up and this is called brain entrainment.

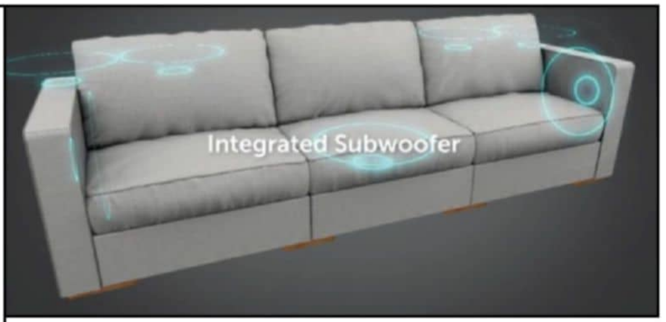
The following brain pattern/frequencies and are generally accepted as fact:

- Delta pattern: Binaural beats in the delta pattern operate at a frequency of 0.5Hz to 4Hz, which also typically indicates a dreamless sleep. In one study, people who received a delta pattern frequency during sleep entered a deeper stage of sleep, according to electroencephalogram (EEG) brain scan results.
- Theta pattern: Practitioners set binaural beats in the theta pattern to a frequency of 4Hz to 7Hz. Theta patterns contribute to improved meditation, creativity, and sleep and REM rapid eye movement phase.
- Alpha pattern: Binaural beats in the alpha pattern are at 7Hz to 13Hz and may encourage relaxation.
- Beta pattern: Binaural beats in the beta pattern are at 13Hz to 30Hz. This frequency range may help promote concentration and alertness, but may increase anxiety at the higher end of the range.
- Gamma pattern: This frequency pattern accounts for a range of 30Hz to 50Hz. These frequencies may promote maintenance of arousal while a person is awake.

With many therapy devices, there is an unstated assumption that haptics (and audible sounds) will induce the user's brain into shifting into the target awareness, mood, and emotional state, but this is still uncharted territory. From the devices I have checked out, all this stuff definitely has some positive impact on your state of mind. Maybe I just need a break from time to time, but really that is what this stuff is all about.

Haptic for Music and Movies

In some haptics products, aside from "new age" music, other program material (e.g., movies) can be enhanced by the immersive multi-sensory experience. Dolby surround formats (and most other surround sound movie formats), have a dedicated "boom channel"



The home furnishing brand Lovesac now offers Sactionals StealthTech Sound + Charge with immersive surround sound by Harman Kardon, a complete "under-the-couch" sound and haptics solution with convenient wireless charging, seamlessly embedded and hidden inside the adaptable Sactionals platform.

(LFE) and most audio systems have a subwoofer output. Voice through haptics mostly does not work well and the effects channel does not include the dialog from the program sent to the subwoofers (and tactile devices). Yet I should mention bone conduction for smart glasses, a number of flat screen TVs, and other applications actually accomplish human voice quite well, but that is a topic for another article.

Monster developed a bass shaker "action couch" many years ago, and there is the D-BOX motion system that is built-into couches in cost-no-object home theater systems. These are firmly in the more "shock and awe" home entertainment and gaming side of haptics, and now joined with Harman Kardon's collaboration with home furnishing brand Lovesac. Their Sactionals StealthTech Sound System includes two couch-embedded front- and rear-firing Harman Kardon speakers, a subwoofer that integrates into the seat frame and a center channel, all completely hidden from view. Users can expand the system with up to four satellite speakers for additional surround sound and satellite subwoofers. The sound properties are optimized to the density of the chosen fabric covers, allowing sound to pass through fabric and upholstery. To quote Lovesac "whether you're watching a movie as a family, meditating alone, or watching the game with friends, the System adapts to your lifestyle without sacrificing style and comfort." On demonstration in Lovesac and select Best Buy locations, with a retail price of $3,700.

Resonate's Vibrational Chair and Ottoman

Resonate has taken the haptic furniture concept to a place that evokes the phrase made popular through its use in the title sequence

Resonate combines traditional meditation techniques with the proven benefits of vibration, sound, and light to help us relax and improve mental health. A Resonate system comes with a chair and matching ottoman, custom light glasses, headphones, and a subscription to a proprietary content library.

of *Star Trek*—"where no man has gone before." Using a recliner format, Resonate has created effortless Modern Meditation. Most of us are deeply grounded and very much in the moment of the here and now and extracting ourselves from the daily grind can be a lost skill. Resonate is on mission to make meditation more accessible. Resonate uses the science of neural entrainment to transform meditation from hard to easy, helping access to deeper levels of consciousness. Having lived with a development prototype, I can say it works. More than some audio gear in a couch, and quite an impressive creation.

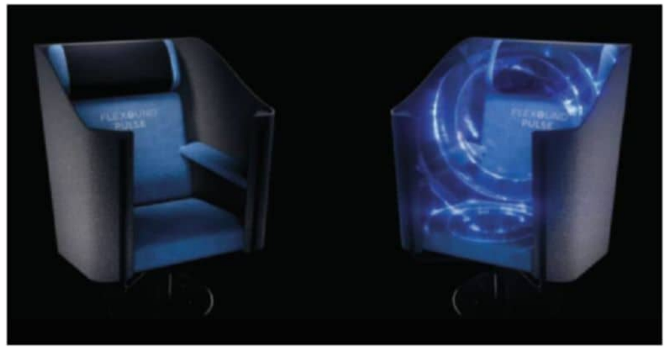
By combing sound, vibration, and visual senses, a truly immersive experience lets you get away from reality for a while. It is so powerful that in my first encounter I could not estimate the duration of the experience—it could have been 5 minutes or the entire afternoon (spoiler—it was 20 minutes).

The haptics recliner is combined with LAYER, a high-tech LED matrix display—sort of a visual analogy of binaural beats. The LED light matrix translates videos of the natural world into an ever-changing sequence of biomorphic patterns visible through closed eyes and proven to activate the brain's frequency-following response. All this and traditional meditation techniques to change the frequency and pattern of thoughts. Users report a 51% decrease in stress and anxiety levels after one 20-minute resonate session. Having written this article just after coming back from the dentist, this would have enhanced that experience (along with nitrous oxide) but perhaps too "new-age" for my dentist. Shipping about the time you read this.

Flexound Immersive Pulse Seats

Flexound Augmented Audio, a company from Finland that fully explores the combined potential of haptics and audio in consumer products, is also developing multisensory solutions for cinema, gaming, and therapy. After exploring the technology in commercial movie theaters, Flexound has now introduced the Pulse seat that offers an affordable and captivating home cinema experience, scaling easily to large multi-seat rooms.

The company created earlier products that combine full-range, high-quality audio with gentle physical sound wave vibration, to add the sensation of touch, complementing any listening experience, be it music, games, television, streaming, meditation, movies, VR, or any other sound content. Originally designed and engineered in Finland to help autistic children, the Flexound technology can add value in many industries. The company first explored the potential with its pilot consumer product HUMU Augmented Audio



The new Pulse seat from Flexound Augmented Audio is a cinema seat that is able to convert frequencies up to 500Hz into haptic vibrations, and even up to 1kHz in direct contact with hands and skin.

Cushion, which enables users to hear and feel their favorite content without disturbing others around them. HUMU combines Bluetooth audio with haptics transducers to generate an immersive personal soundsphere with amazing bass that can be carried anywhere. The HUMU pillow is commercially available now, and users are discovering the possibilities for all kinds of home entertainment, from playing games to enhancing audio in home offices, while also using it for mediation and relaxation.

Our sense of touch is amazingly sensitive to sound and has been consistently underused in high-quality audio, except for low frequency effects in movies. Flexound aims to make audio a whole-body experience by enhancing vibrations through foam, providing a soft, nearfield listening experience. The company's patented technology can be embedded into seats, cushions and anything that is soft and in contact with the body.

The new Pulse seat from Flexound is a cinema concept that is able to convert frequencies up to 500Hz into haptic vibrations, and even up to 1kHz in direct contact with hands and skin. By enhancing the sound with haptics, the sound reproduced by speakers doesn't need as many decibels to achieve clarity of speech, and dramatic scores and sound effects won't mask dialog, even during the most chaotic action movie sequences. And because Pulse does not rely on far-field listening or filling a room with decibels (e.g., traditional loudspeakers), sound leakage from room to room is considerably less, enabling a fully functioning cinema or gaming station to be placed in any type of room, at home or in an auditorium.

Essentially, Flexound designed a comfortable and safe seat that is able to reproduce the full range surround sound cinema experience with its integrated, patented augmented audio technology which explores the potential of haptics transducers. Users feel and hear every little note and sound in their own personal sound-sphere. The system supports all existing sound content and formats, enhancing the emotions created by the content, and also makes it accessible to more people, such as the hearing impaired.

Dreampad

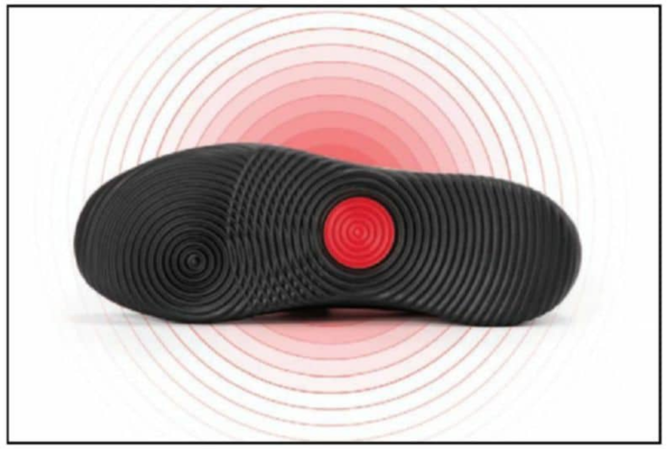
The Aurras is a new product from Dreampad, which has been delivering high-quality music pillows for relaxation and sleep therapy since 2014. The Dreampad pillow was developed from a discovery that bone conduction sound produces a relaxation response in the body and the initial application to help autistic children. Based on this therapeutic insight, the vibrational music technology was embedded in pillows, to create a relaxing and restful bedtime sound experience using bone conduction technology, which permeates the entire pillow but not beyond, not bothering the person next to you in bed.

Dreampad Parras sends soothing sounds and gentle vibrations through the pillow via Bluetooth.

The Aurras is made specifically to be enjoyed in bed, without cumbersome headphones or earbuds, and without disturbing others. With a Bluetooth amp with integrated equalizer, the Aurras slips under the pillow and lets you listen to the news, movies, podcast, new age music or whatever. Research shows that the Dreampad's unique technology, combined with specially engineered program material, can reduce stress and anxiety, helping faster time to sleep and stay asleep.

DropLabs Haptic Sports Shoes

We have previously explored examples of haptic belts, vests, and headphones but how about sports shoes? DropLabs, a company from Los Angeles, CA, has Bluetooth water-resistant sneakers that shake your feet with bass (happy feet!). DropLabs sound-activated footwear uses low-frequency information to extend entertainment experiences. The technology converts audio into vibrations that are felt from your feet through your entire body, transforming music, movies, games (or binaural beats) into immersive, live experiences.



DropLabs' patented technology uses a haptic driver embedded in the midsole of the shoe, sending audio, in stereo, from any connected device to the user's feet, and creating full-body bass audio.

DropLabs's products look like standard sneakers or training shoes, but are equipped with sound generating technology and Bluetooth, embedded enhanced audio controls, multicolor LEDs plus a few surprises. They produce tactile vibrations accurately and offer both iOS and Android apps that allow you to toggle the bass intensity. DropLabs uses a haptic driver embedded in the midsole of the shoe, creating full-body bass audio which simulates the energy of a live event and immerses users in music, gaming, VR content, and more. The wireless audio transmission is effectively converted into a haptic extension of whatever the user is hearing—music beats, movie, or gaming effects.

Apollo Neuroscience

The Apollo Neuro is a wearable that provides real-time stress control for better sleep, focus, energy, and calmness through haptic sensations to guide your moods and energy levels. All sound is vibration, and while we don't interpret Apollo's vibrations as music, it affects how we feel nonetheless.

The device is in the form-factor of a smartwatch and includes a band for the wrist and another one for ankle-worn use, and I found both comfortable, but for me the wrist was most effective. Dr. David Rabin, a board-certified psychiatrist and neuroscientist, developed the device while studying at the University of Pittsburgh Medical Center, followed by five years of testing and clinical studies.

A smartphone app is supplied for automatic Bluetooth pairing, basic instructions, and selection of the mood program, including time and strength. Seven modes range from Energy and Wake Up, Social and Open, Clear and Focused, Rebuild and Recover, Meditation and Mindfulness, Relax and Unwind, and Sleep and Renew. Each mood program cycles through various stages, so it isn't repeating the same pattern for the entire duration.

Participants using the wearable measured better cognitive performance and heart-rate variability (HRV) than two control groups—one using placebo vibrations and another not using any wearable tech. One user describes the experience like having a purring cat on her lap. Other Apollo Neuroscience articles claim improved HRV and lower heart rates for subjects in performance or other stressful situations.

The Neuro from Apollo Neuroscience is a wearable that provides real-time stress control through vibration, paired with a smartphone app.

The Fisher Wallace Stimulator is a neurostimulation device that is intended to reduce depression, anxiety, and insomnia by stimulating the brain to produce serotonin with a gentle electrical current.

Direct Brain Stimulation?

Aside from haptics and acoustical sensations, there are adjunct techniques such as cranial electrotherapy simulation. A relevant example is the stimulator from Fisher Wallace. Two spongy electrodes are applied to either side of your forehead and using a low-level electrical pulse to stimulate the brain to produce serotonin and other neurochemicals required for healthy mood and sleep, while also entraining a calm brainwave state and modulating the "fight or flee" nervous system.

To treat depression and anxiety, patients wear the device for 20 minutes, twice per day—once in the morning and once before bed. To treat insomnia only, the device may be used once before bedtime. Beyond just an electronic "happy pill," the Fisher Wallace Stimulator is also being used to treat Parkinson's disease.

Parkinson's disease is a progressive nervous system disorder that affects movement. In the early stages of Parkinson's disease, the face may show little or no expression, the arms may not swing when walking and speech becomes soft or slurred. The symptoms of Parkinson's disease worsen as the condition progresses over time. Although Parkinson's disease can't be cured yet, medications significantly improve symptoms. A more drastic solution is surgery in the form of Deep Brain Stimulation (DBS), which is being used to regulate certain regions of the brain and improve symptoms. A surgeon places thin metal wires in the brain; these wires send electrical pulses to the brain to help control some motor symptoms. While transformative for some, DBS is not for everyone. For now, the Fisher Wallace has been found effective in treatment of Parkinson's disease without the need for surgery and "brain implants."

Gaming Vests and Wearables

The Aura Interactor was the first commercially available wearable force-feedback vest, released in 1994. The Interactor monitored the audio signal and used Aura's patented electromagnetic actuator to produce vibrations that can represent such actions as a punch or kick. It was compatible with the Super Nintendo, Sega Genesis and Sega CD consoles. Back then, the gaming developers and the game hardware lacked effects or bass tracks for tactile devices, which limited performance from a user's perspective. Today, we have haptic gaming vests and belts from companies such as Woojer.

Woojer's Belt Edge and Vest Edge are wearables, with a tactile transducer, their core technology. The patented Oscillating Frame Actuators (Osci) deliver a combination of tactile sensation, with precise and realistic haptic technology, in the required size, weight, and

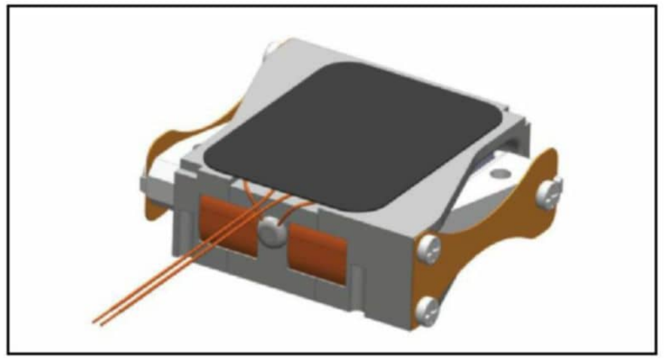
power consumption. The actuator is designed without a resonator and generates the haptic sensation while maintaining complete silence. A closed-loop metal frame forms the load-bearing structure of oscillating permanent magnets. The frictionless motion guarantees precise harmonic reproduction and reduces distortion. To fine-tune the haptic sensation, the frequency response is tailored by modifying the cross section of the frame. The design utilizes an "I" shaped core as the transducer's stator and the human body as the resonator. This combination minimizes the number of elements. Operating band is for frequencies up to 200Hz, including subsonic frequencies that are below the threshold of the human ear (1Hz to 20Hz).

Placement of the actuators on body hotspots convince the brain, through the principle of Perceptual Inference (auto-completion), that the entire body is exposed to the high acoustic energy. Two tactile products have been developed, a belt and a vest.

Also targeting gaming applications, headphones are now being "enhanced" with haptic systems creating a more "portable" and convenient solution that doesn't requiring putting on full-body armor. Gaming headphones from Razer (e.g., the Nari Ultimate) have built-in wide-band voice coil actuators from Lofelt, designed to deliver a high-definition haptic experience. Not to be confused with the larger bass shakers, the Lofelt L5 actuator is about the size of a wristwatch at 17mm × 20.5mm × 6.2mm and weighing 6 grams. It offers a rich and powerful vibrotactile effect and can be driven using standard audio signals from low-power audio amplifier ICs with a haptic sensitivity range of 45Hz to 250Hz. Within the recommended conditions, the actuator is capable of operating continuously for hours—ideal for enhancing immersive experiences in gaming and VR/AR environments. Additionally, the L5 actuator works in combination with the Lofelt Wave real-time firmware and full haptic effects software suite to provide haptic performance optimization.

Another company successfully exploring haptics in headphones is lifestyle audio brand Skullcandy. Always trying to offer something unique, Skullcandy currently offers multiple Crusher headphones, which feature Skullcandy's patented Sensory Haptic Bass. The Sensory Haptic Bass feature has turned into a defining attribute of the Crusher headphones, becoming widely recognized as a unique audio experience, appreciated beyond gaming. The latest Crusher models introduced in 2020 refine the haptics platform with more multi-dimensional vibration patterns across a wider range of bass frequencies. The latest models also combining Sensory Haptic Bass with active noise canceling and personalization from Audiodo.

The Woojer gaming vest uses a patented Oscillating Frame Actuators (Osci) technology that delivers a combination of tactile sensation and realistic haptic technology.



The Lofelt L5 actuator is a wide-band voice coil actuator designed to offer a rich and powerful vibrotactile effect and can be driven using standard audio signals from low-power audio amplifier ICs.

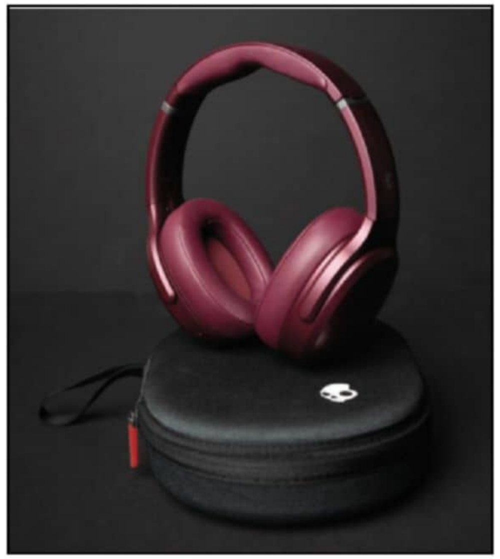
The flagship Razer Nari Ultimate gaming headphones featuring Lofelt's actuators and HyperSense technology transform audio frequencies into haptic effects with different intensities and precise localization.

Resource Companies

Apollo Neuroscience, Inc. | www.apolloneuro.com
Dreampad | www.dreampadsleep.com
DropLabs | www.droplabs.com
Fisher Wallace Laboratories | www.fisherwallace.com
Flexound | www.flexound.com
Lofelt GmbH |www.lofelt.com
Lovesac | www.lovesac.com
Resonate | www.weresonate.com
Skullcandy | www.skullcandy.com
Woojer, Inc. | www.woojer.com

AAC Technologies is another established vendor of microspeakers for smartphones that long ago expanded to the vibration buzzer and more recently for exciters for all-screen devices. The company is developing a range of improved screen actuation technologies to produce audio that is driven by the display itself, eliminating the need for a (earphone) receiver opening—which allows for a larger screen and reduced borders. Directly in haptics, AAC is developing new form factor actuators with increased acceleration. The most common applications is virtual keys (virtual touch), featured in a wide range of devices from smartphones to smart home devices. AAC's actuator portfolio and close collaboration with sensing partners, coupled with AAC's advanced 3D manufacturing of market-leading glass, creates a unique and distinct touch experience.

The next step in this field is a new generation of augmentation haptic interface systems, which enable connecting users with remote—or even virtual—experiences. More recently, the whole concept of the Metaverse promises to expand on attempts to connect the physical and the virtual worlds—or what is being described as "persistent online 3D virtual environments" and augmented reality. Haptics systems and actuators will be a natural and vital extension for all those concepts.



Skullcandy Crusher headphones feature Sensory Bass with new, patented drivers that enable users to hear and feel the music, adjusted according to preferences.

Haptics Technology and Trends

IDTechEx Research published its report "Haptics 2021-2031: Technologies, Market & Players" discussing the general trends in the haptics industry and the provision of haptic feedback within products today. According to the report, the industry segment will be worth nearly $5 billion in 2025. The report covers each major type of haptic technology, from electromagnetic actuators to gaming, wearables, consumer electronics, and automotive haptics.

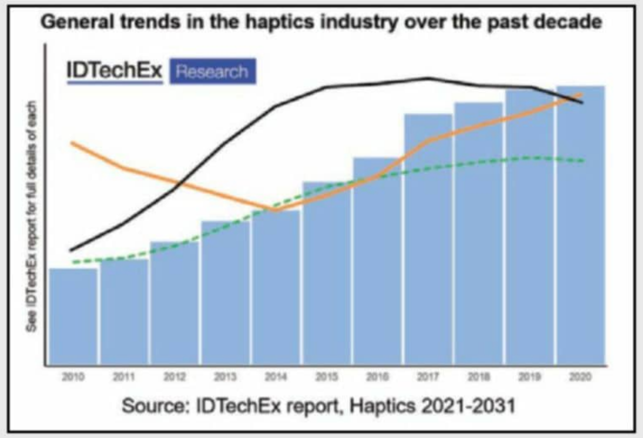
As companies compete to release the best new electronic devices, differentiation from the competition comes from the different features and components that a device can offer. Investment in these different components differs over time; in any particular year or product cycle, a player can differentiate from the increasingly busy crowd by winning with a specific component or feature development, enabling it to stand out. This could be an advanced camera or image recognition processor, 5G connectivity, or a new display or interface.

Haptics is one of the more unsung heroes among these components, but while it rarely makes headlines, it contributes significantly to the overall experience of using devices. Investment and development in haptics is accelerating, and the supply chain providing the actuators, drivers and software behind these key components is evolving at a faster pace every year.

The smartphone dominates the market for haptics in terms of volume. However, it is in other industry sectors and products where newer, more advanced haptics often find their feet first. Some of the most advanced haptics have been developed for military use, for controlling high-end industrial equipment, or the most advanced surgical robotic assistants. Haptics are being explored in vehicular interfaces, replacing mechanical buttons and using the sense of touch for a safer, more reliable interface to the driver. It is also fundamental to VR, AR, and MR, and console gaming, where controllers deploy haptics to enhance player immersion, as exemplified with Sony's significant new investment into haptics within the DualSense controllers for PlayStation 5.

With new haptics being integrated into products spanning several industry verticals, many options for new technologies are approaching maturity, and IDTechEx predicts production efforts for the next generation of actuators for key markets such as smartphones to be ramped up within the next 1 to 2 years. "New commercialization drives new interest into the next generation of haptics, fueling additional investment into parallel technologies, which may serve as alternatives or future options for these lucrative consumer product markets. It also drives enhanced user expectations in haptic feedback, which impacts other haptics markets, from industrial applications to automotive interfaces," the report states.

www.IDTechEx.com/Haptics



The latest IDTechEx report is supported by more than 5 years of covering that specific side of the industry, including historic market data from 2010 and supporting databases of companies and products in the haptics space. (Source: IDTechEx report, "Haptics 2021-2031: Technologies, Market & Players")

Sound Solutions

World's First Actuator Loudspeaker for Stereo Sound

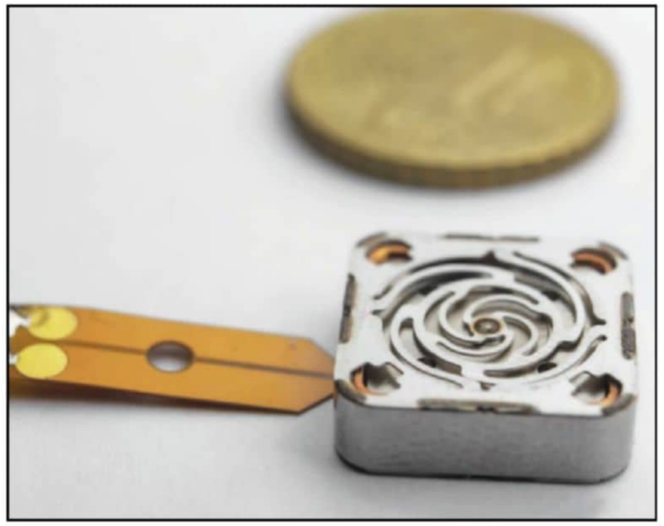
The micro-acoustics experts at Sound Solutions have developed a compact actuator that is able to ensure stereo sound quality when used to transform the display of increasingly thin smartphones and tablets into a loudspeaker. According to the Vienna-based company, the new "Singing Display" actuator is ideal for space-saving, waterproof designs, extending the response beyond what is possible with conventional speaker technologies, by being applied directly to the back of the large tablet or smartphone screen surfaces.

Compared to dynamic speakers, the advantage of this technology is in the significantly reduced volume/space, which is essential for smartphones and tablets that are becoming thinner and thinner. In addition, no holes/openings for sound outlet are required in the housing, which means that new designs using full display solutions and benefiting from complete water tightness can be implemented/realized.

Other actuator components for sound generation have already been developed in recent years, and some are used in various commercially available mobile phones from different providers. However, in all these devices until now, the components could only be used for low-volume reproduction during communications, when the smartphone was placed close to the ear. The acoustic applications are also limited to mono audio.

The new actuator component now available from Sound Solutions International (SSI), is able to provide full, hands-free, high-quality sound reproduction through the display, which means it can replace the top speaker used for communication and at the same time enable stereo sound reproduction. Another major benefit lies in the fact that each stereo audio component has an extremely small footprint (13.6mm × 13.6mm). In addition, the innovative solution can also enable haptic feedback as an additional feature for free—this enables OEMs to save costs by also eliminating the need for an additional haptic engine and save even more space.

www.sound-solutions.com



MSE Audio

SolidDrive Surface-Mount Actuators with 70V Transformers for Sound Masking and Background Music

MSE Audio announced three new models of SolidDrive surface-mount actuators with transformers. These actuators convert audio signals into vibrations that transfer into solid surfaces through direct contact and work well for sound masking and background music applications. Available in drywall (SD1-T), wood (SD1-sm-T), and glass (SD1G-T) versions, each piece is designed for one channel of audio. All models are now shipping globally.

SolidDrive actuators use patent-pending dual-voice coil technology in an aluminum enclosure to transfer acoustic energy to virtually any substrate. Featuring high-powered neodymium magnets and dual symmetrically opposed motors, these convert audio signals into vibrations that transfer into any solid surface through direct contact. The result is sound that radiates from the entire surface with a full-frequency response of 70Hz to 15kHz. The symmetrically opposed neodymium magnets and specifically engineered mounting hardware provide hidden planar-source acoustics and uniformity throughout the listening space. And with the new transformer versions of SolidDrive, the company added another deployment option for integrators.

Optional accessories are available to support various installation needs, including a Titanium Heat Sink (SD1-HS-TI), recommended for hot weather locations and in applications when SolidDrive is attached to a small surface area, and 12" (SD1-BR12), 16" (SD1-BR16), and 24" (SD1-BR24) On-Center Bracket Assemblies. These assemblies have patented balanced cantilever spring brackets and are required for installations of the SD1 behind drywall. The mounting bracket eliminates stress on the glue bond by balancing the SD1 to eliminate torque and by holding it against the drywall with proper amount of pressure.

Also available are Passive Equalization Modules for drywall (PEM-DW), wood (PEM-W) and glass (PEM-G), designed to smooth out the frequency response and optimize the sound of the SD1, SD1g, and SD1sm for any type of surface to which they will be attached. One module is needed for each SolidDrive unit.

The SD-1 Desktop Kit-250 is the ideal way to see how the SolidDrive works on any surface—including but not limited to glass, metal, wood, ceramics, laminates, and composites. The SolidDrive unit in the kit has a special silicone backing so it will work with virtually any surface without scratching or leaving a mark. The kit includes one SD1 desktop unit, DC power adapter, cables, and SD-250 250-watt mini amp with SolidDrive EQ curves in a convenient carrying case.

www.soliddrive.com
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Cirrus Logic

Advanced Haptic and Sensing IC Solutions for Richer, Immersive User Experiences

Already a leader in haptic technology for smartphones, Cirrus Logic expanded its haptics and sensing expertise with the new Cirrus Logic CS40L25 family of boosted haptic drivers. The new ICs enable OEMs to create customized user experiences beyond the single-action response of today's mechanical buttons.

Cirrus Logic's haptic solutions feature low-latency, boost voltage and closed-loop control algorithms that enable customized waveforms to trigger a variety of fast, unique haptic feedback responses. The new haptic solutions from Cirrus Logic can help create context-aware "virtual" buttons for almost any surface. By eliminating mechanical buttons, product designers can create cleaner, sleeker industrial design aesthetics with no button "bumps." Many smartphone designers are leading this behind-the-screen design evolution to increase haptic feedback solutions by replacing peripheral button functions. Automotive controls, PCs, wearables, and game controllers are also moving beyond traditional button interfaces to incorporate non-mechanical haptic feedback.

Cirrus Logic's CS40L25 products integrate a high-performance haptic driver, a digital signal processor, and a boost converter. The devices are resonance-aware, drive high-performance linear resonant actuators (LRAs) and voice coil motors (VCMs), and enhance user experiences by supporting unique/pre-stored haptic waveforms. Ultra-low latency provides real-time control of the haptic motor. This provides users with a more immediate sensation or response. Closed-loop algorithms maximize LRA effectiveness and enable strong and consistent haptics with a crisper, less "buzzy" effect.

In addition, Cirrus Logic is offering a new-generation haptic solution that integrates force sensing and the haptic driver. The single-chip device is anticipated to improve performance, reduce power consumption and simplify system design with up to a 50% reduction in the overall footprint of a smartphone haptic subsystem.

www.cirrus.com/products/CS40L25

A High-Quality MM Phono, Line, and Headphone Amplifier

By
George Ntanavaras

Photo 1: My completed PhL-100 works as a phono preamplifier for moving magnet catridges and a line/headphone amplifier.

In this project article, the author describes the design and construction of a phono preamplifier for moving magnet (MM) cartridges and a line/headphone amplifier with two additional inputs and volume control, which he calls the PhL-100.

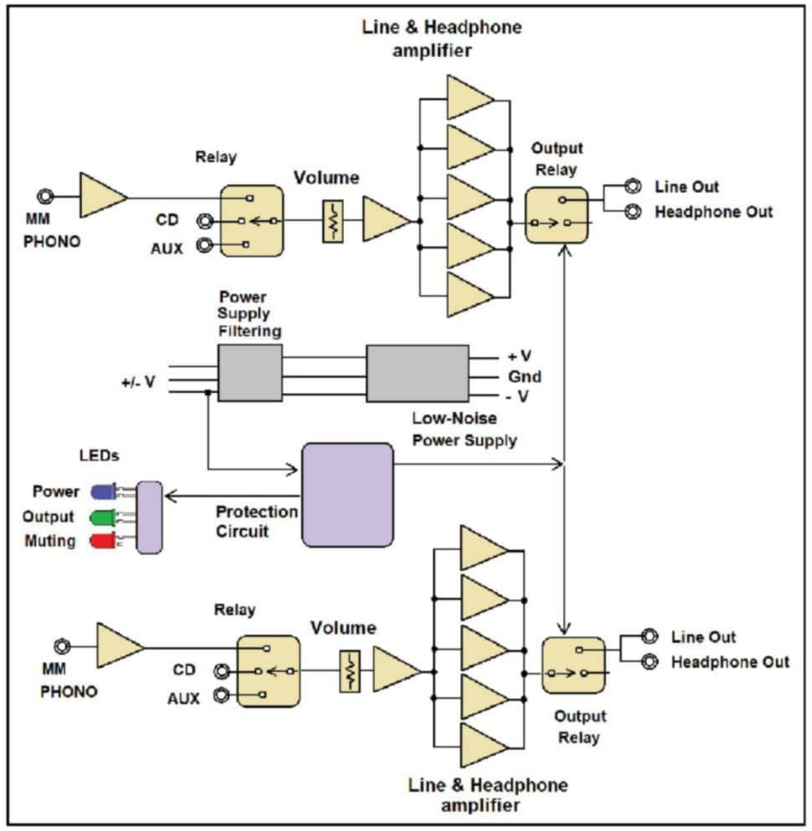

Figure 1: The simplified block diagram shows all the parts of the PhL-100 device, including a protection circuit.

The completed PhL-100 device is shown in **Photo 1**. To begin, all the amplification stages in my PhL-100 design are based on top-quality audio op-amps from Texas Instruments (TI), where I selected the OPA1642 for the phono stage and OPA1612 for the line/headphone stage. Almost all the components of the device are contained on a single PCB for easy construction.

Figure 1 shows the simplified block diagram of the PhL-100 device with all the parts of the preamplifier—the MM phono stage, the line and headphone amplifier, the protection circuit with the indication LEDs, and the power supply.

The Phono Preamplifier Circuit

The detailed block diagram of **Figure 2** shows that each channel consists of two gain stages and a DC servo circuit that eliminates the output offset. The first stage buffers the cartridge and provides a gain of 21.3dB. A passive circuit follows with the RIAA equalization time constant at 75μsec. The second stage provides additional gain and the 318μsec and the 3180μsec time constants of the RIAA curve.

The electronic schematic is shown in **Figure 3**. A detailed description of the operation follows, which refers only to the Right channel, since the other is identical. Resistor R2 and capacitor C5 adjust the load for the moving magnet (MM) cartridge. Both should have the recommended values by the cartridge manufacturer. The schematic shows the typical values of 47kΩ and 100pF.

The first stage is a non-inverting amplifier based on an OPA1642. This is a JFET-input, ultra-low distortion, low-noise, dual op-amp fully specified for audio applications. The gain is 21.3dB and it is defined by resistors R3 and R4, which have low values for minimum noise. This gain is suitable for cartridges with a nominal output about 5mV @ 1kHz, 5cm/sec. If the cartridge has a different nominal output, the gain should be adjusted accordingly by changing the value of R4. The capacitor C4 with R3 give a high frequencies roll-off at about 300kHz.

Resistors R1, R7 and capacitor C14 form the time constant of the RIAA equalization at 75μsec (2122Hz). I used high accuracy resistors with 0.1% tolerance and polypropylene capacitor with a tolerance of 2% to minimize the deviations of the circuit from the RIAA curve.

The next op-amp stage is based again on the OPA1642 op-amp and forms the time constants of the RIAA equalization at 3180μsec and 318μsec with the resistors R5, R6, R8, R9, R10 and the capacitor C8. Here again I used high-accuracy components.

The total gain of the phono preamplifier is 42dB at 1kHz. Resistor R16 buffers the output of the phono stage before it is connected to the Line preamplifier.

The circuit around IC3 is a non inverting DC servo integrator, which keeps the output of the RIAA preamplifier very close to 0VDC. The use of the servo circuit eliminates the use of a capacitor in the feedback loop of the amplifier. The Analog Device AD706J is used for IC3, which is a dual, low power, bipolar op-amp with 100μV input offset, and input bias current < 100pA with low drift over temperature. The output voltage of IC3 through resistor R48 adjusts the voltage of the inverting input of the second amplification stage in order to eliminate any DC offsets at the output.

The Line/Headphone Amplifier Circuit

The electronic schematic of the line and headphone amplifier is shown in **Figure 4**. The selection of the input signal of the preamplifier is performed by two high-quality relays per channel, K1 and K2 from Panasonic, type TXS2-12V-1 with AgPd contacts, suitable for use in low level load analog circuits. Across each relay coil, protection diodes are used (D16-D17). Each external input has one dedicated double pole relay that connects or disconnects completely both the positive and the negative part of the signal from the preamplifier circuitry.

Resistor R17 or R29 with capacitor C3 form a low-pass filter at about 230kHz, (assuming that the previous stage's output impedance is not greater than 200Ω) to attenuate any electromagnetic interference (EMI) that could enter to the circuit. Next, there is the connector for the volume control potentiometer, which is placed on the device's front panel. The volume control's position at the input protects the preamplifier from any possible overload that could caused by high level input signals. I used a rotary stereo logarithmic potentiometer from Alps (type RK27112-LOG50k). The capacitor C11 is included for the DC protection of the line amplifier circuitry and it is the only capacitor used in the audio signal path. Resistor R33 needs to ensure a DC path for the op-amp proper operation.

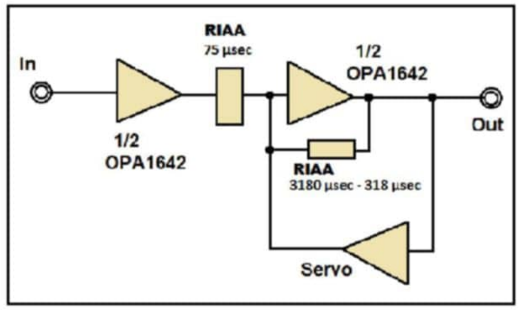

Figure 2: The detailed block diagram of the phono stage shows that each channel consists of two gain stages and a DC servo circuit that eliminates the output offset.

The amplification stage is built around IC5A, which is an OPA1612 connected as a non-inverting op-amp circuit. The OPA1612 is a bipolar, ultra-low noise (1.1nV/√Hz at 1kHz) op-amp with an ultra-low distortion (0.000015% at 1kHz). It offers a rail-to-rail output swing with a high slew rate of 27V/μs at 2kΩ load, which increases headroom and maximizes dynamic range. It is also unity gain stable and has a high output drive capability.

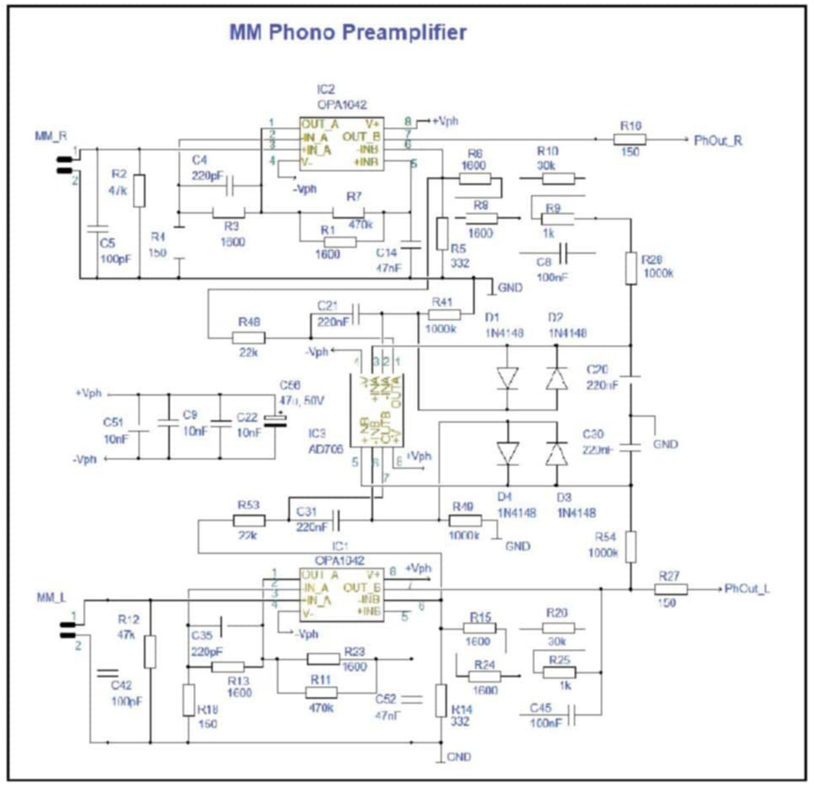

Figure 3: This is the electronic schematic for the phono preamplifier.

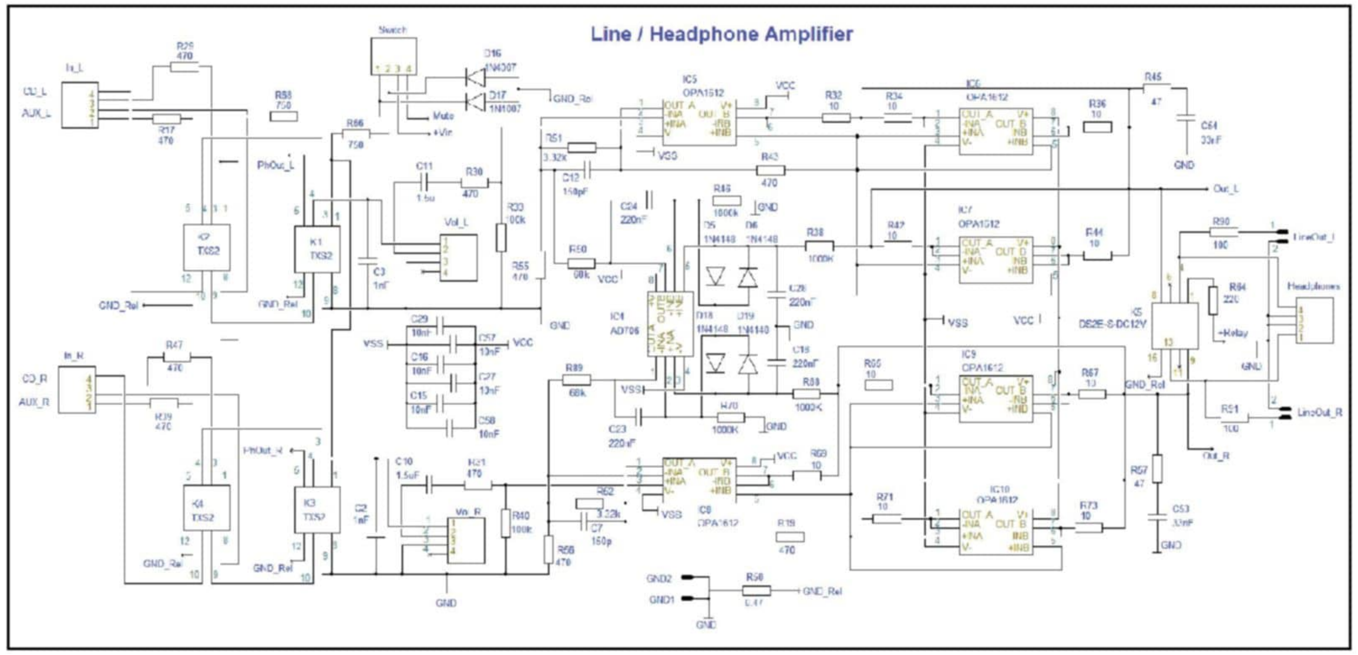

Figure 4: Here is the electronic schematic for the line and headphone amplifier.

The gain defined by resistors R51 and R55 is set at x8 (18.1dB). Resistor R55 has a low value for minimum noise contribution to the circuit. The value of capacitor C12 was selected so that the roll-off of the high-frequency response will be at about 320kHz.

Resistor R43 buffers the output of the IC5A from the next stages that consist of five OPA1612 op-amps (IC5B, IC6A, IC6B, IC7A, and IC7B) operating as voltage followers with their outputs connected in parallel by 10Ω resistors. The total maximum output current capability as I measured is about 200mArms. The components R45 and C54 at the output help keep the high-frequency stability when driving long cables or difficult loads.

Resistor R90 buffers the output of the line preamplifier from capacitive loads in case long cables are connected from the preamplifier to the power amplifier.

The outputs are isolated from the circuit through relays, which close with a short delay during the power up and disconnect very quickly during power down. Also they can be disconnected manually when the MUTE switch is energized. The circuit around IC4 is a DC servo integrator similar to the one used in the phono stage to keep the output of the amplifier very close to 0VDC.

The Power Supply Circuit

The PhL-100 needs an external regulated power supply between ±12.6VDC to ±15VDC, with a minimum current capability of ±0.5 A. It could operate with lower voltages if the Zener diode Z2 of the protection circuit is changed to a lower voltage.

I used a low noise external power supply based on capacitor multipliers and LM317 to LM337 voltage regulators, similar to the one I used for my headphone amplifier published in *audioXpress*, June 2013 (see Resources). The voltage outputs of this power supply were ±12.6VDC.

I also used an external desk-top switching power supply for the PhL-100. It has a ±15VDC output with 5% total regulation and a minimum switching frequency at 24kHz, which is outside of the audio bandwidth. With both power supplies the PhL-100 operated very well without any noticeable difference.

Through a 5-pin DIN connector on the back panel of the PhL-100, the voltage goes to the power connector on the PCB. The schematic of the power supply circuit is shown in **Figure 5**.

Fuses F1 and F2 are slow-blow fuses, mounted on the PCB to protect the external power supply from short circuits within the device. FB1, FB2, and FB3 are ferrite beads inductors offering electromagnetic induction (EMI) protection from several megahertz up to gigahertz by suppressing any noise coming from the power supply. Resistors R60, R61, R63, R84 with capacitors C37, C38, C47, C48 form low-pass filters to eliminate any possible noise from the power supply.

The phono circuit is supplied through a capacitor multiplier circuit to further eliminate the noise. Transistors Q1 and Q2 are low noise types and form the capacitor multiplier for the positive side and the negative side, respectively. Resistor R87 and R93 are used to set a 2VDC voltage drop between the collector and the emitter of Q1 and Q2 for proper operation of the circuit.

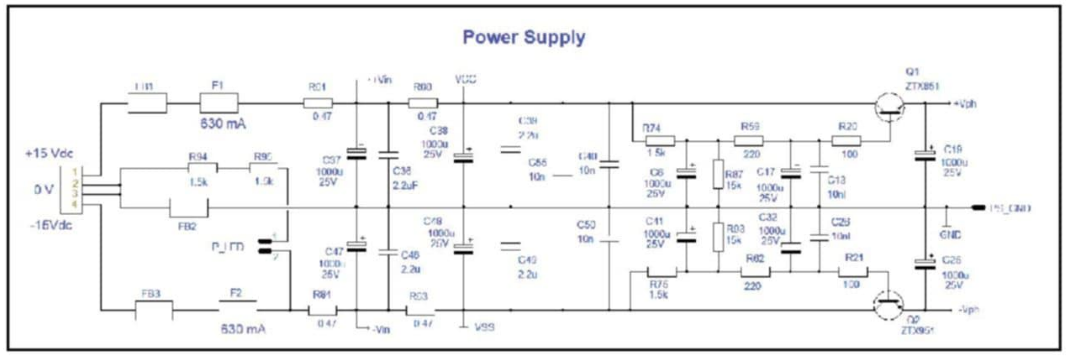

Figure 5: The schematic for the PhL-100's power supply circuit is shown here.

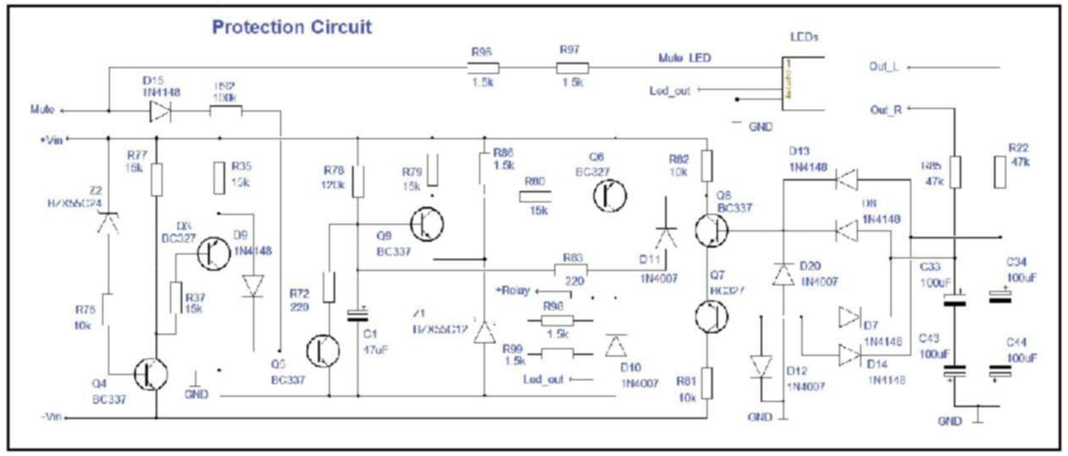

Figure 6: The protection circuit supports several functions.

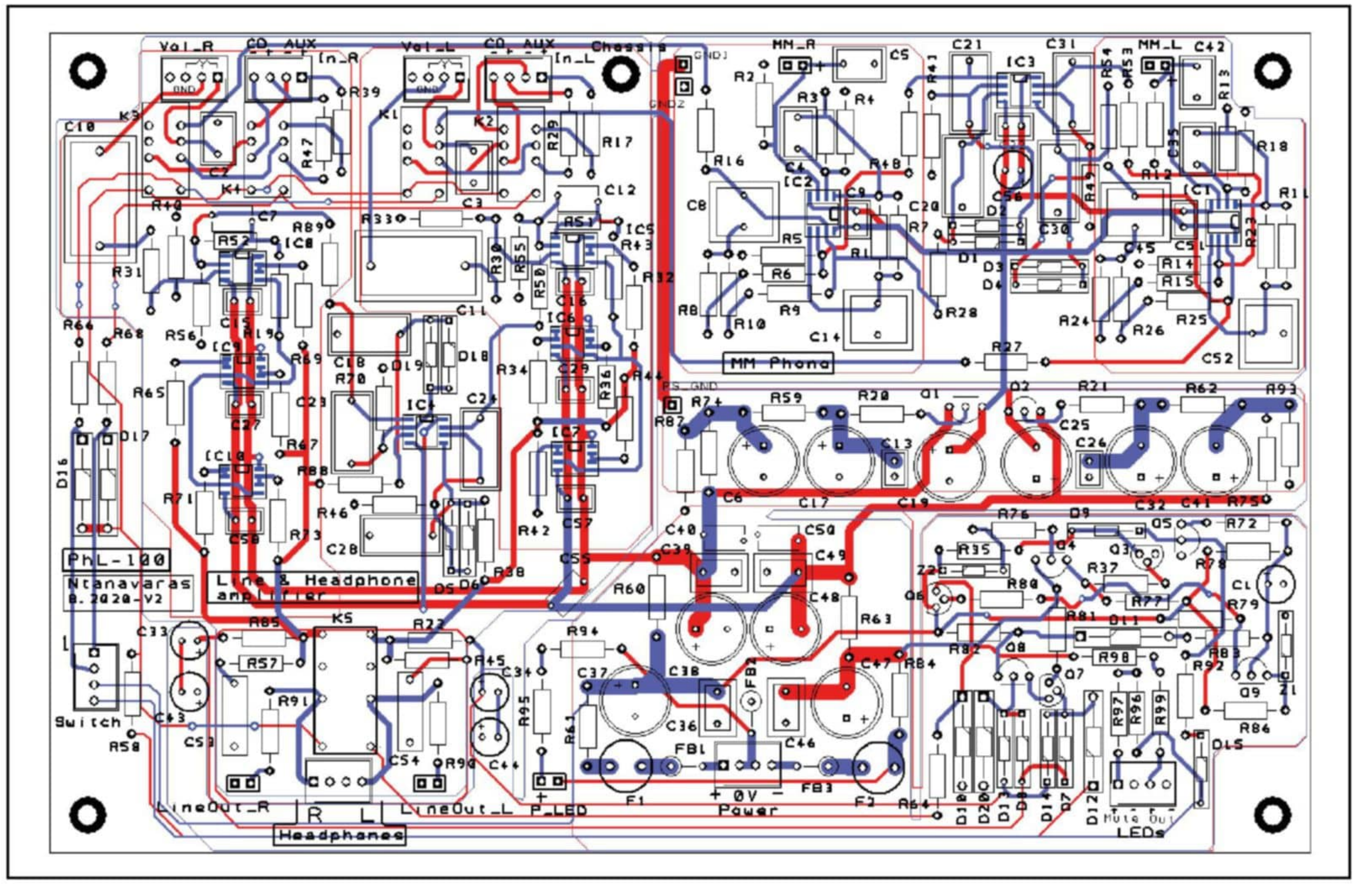

Figure 7: This image details the complete guide assembly for the PCB.

The Protection Circuit

The schematic of the protection circuit is shown in **Figure 6**. The circuit supports the following functions: When the preamplifier is powered on, it connects the output through a relay after a short delay to avoid any transient noise to the following

Parts List for the PhL-100

Part	Value	Description
Capacitors		
C1, C56	47uF, 50V	Electrolytic capacitor, diameter=6.3mm
C2, C3	1nF	Polypropylene, 2.5%
C4, C35	220pF	Polypropylene, 2.5%
C5, C42	see text	According to the MM cartridge specs
C6, C17, C19, C25, C32, C37, C38, C41, C47, C48	1000uF, 25V	Electrolytic capacitor, diameter=10 mm
C7, C12	150pF	Polypropylene film capacitor
C8, C45	100nF	Polypropylene film capacitor
C9, C13, C15, C16, C22, C26, C27, C29, C40, C50, C51, C55, C57, C58	10nF, 50Vdc	Multilayer ceramic COG capacitor
C10, C11	1.5uF	Polypropylene, 3%
C14, C52	47nF	Polypropylene, 2%
C18, C20, C21, C23, C24, C28, C30, C31	220nF	Polypropylene, 5%
C33, C34, C43, C44	100uF, 25V	Electrolytic capacitor, diameter=6.3mm
C36, C39, C46, C49	2.2uF 50V	Ceramic multilayer capacitor, X8L
C53, C54	33nF, 50V	Polyester cap
Diodes		
D1, D2, D3, D4, D5, D6, D7, D8, D9, D13, D14, D15, D18, D19	1N4148	Diode
D10, D11, D12, D16, D17, D20	1N4007	Diode
Z1	BZX55C12	Zener diode
Z2	BZX55C24	Zener diode
Miscellaneous		
K1, K2, K3, K4	TXS2-12V-1	Low signal relay
K5	DS2E-S-DC12V	Low signal relay
Headphones, LEDs, Power, Switch, Vol_L, Vol_R, In_L, In_R	22-23-2041	Connector, 4 pins, Molex KK 254 Series
LineOut_L, LineOut_R, P_Led	22-23-2021	Connector, 2 pins Molex KK 254 Series
F1, F2	630mA	FUSE Radial for PCB Mount
FB1, FB2, FB3	BL02RN2R1M2B	Ferrite beads
Op-Amps		
IC1, IC2	OPA1642AIDR	op-amp
IC3, IC4	AD706JRZ-R	op-amp
IC5, IC6, IC7, IC8, IC9, IC10	OPA1612AIDR	op-amp
Resistors (all 0.25W except as noted)		
R1, R3, R6, R8, R13, R15, R23, R24	1.6kΩ	metal film, 0.1%
R2, R12, R22, R85	47kΩ	metal film, 0.1%
R4, R16, R18, R27	150	metal film, 0.1%
R5, R14	332	metal film, 0.1%
R7, R11	470kΩ	metal film, 0.1%
R9, R25	1kΩ	metal film, 0.1%
R10, R26	30kΩ	metal film, 0.1%
R17, R19, R30, R31, R43, R29, R39, R47, R55, R56	470	metal film, 0.1%
R20, R21, R90, R91	100	metal film, 0.1%
R28, R38, R41, R46, R49, R54, R70, R88	1000kΩ	metal film, 0.1%
R32, R34, R36, R42, R44, R65, R67, R69, R71, R73	10	metal film, 0.1%
R33, R40	100kΩ	metal film, 0.1%
R35, R37, R77, R79, R80	15kΩ	metal film, 1%
R45, R57	47	metal film, 0.1%
R48, R53	22kΩ	metal film, 0.1%
R50, R89	68kΩ	metal film, 0.1%
R51, R52	3.32kΩ	metal film, 0.1%
R58, R60, R61, R63, R84	0.47 Ω	Metal film, 5%, 0.5W
R59, R62, R64, R72, R83	220Ω	metal film, 1%
R66, R68	750Ω	metal film, 1%
R74, R75, R86, R94, R95, R96, R97, R98, R99	1.5kΩ	metal film, 1%
R76, R81, R82	10kΩ	metal film, 1%
R78	120kΩ	metal film, 1%
R87, R93	15kΩ	metal film, 1%
R92	100kΩ	metal film, 1%
Transistors		
Q1	ZTX851	Transistor
Q2	ZTX951	Transistor
Q3, Q6, Q7	BC327	Transistor
Q4, Q5, Q9, Q8	BC337	Transistor

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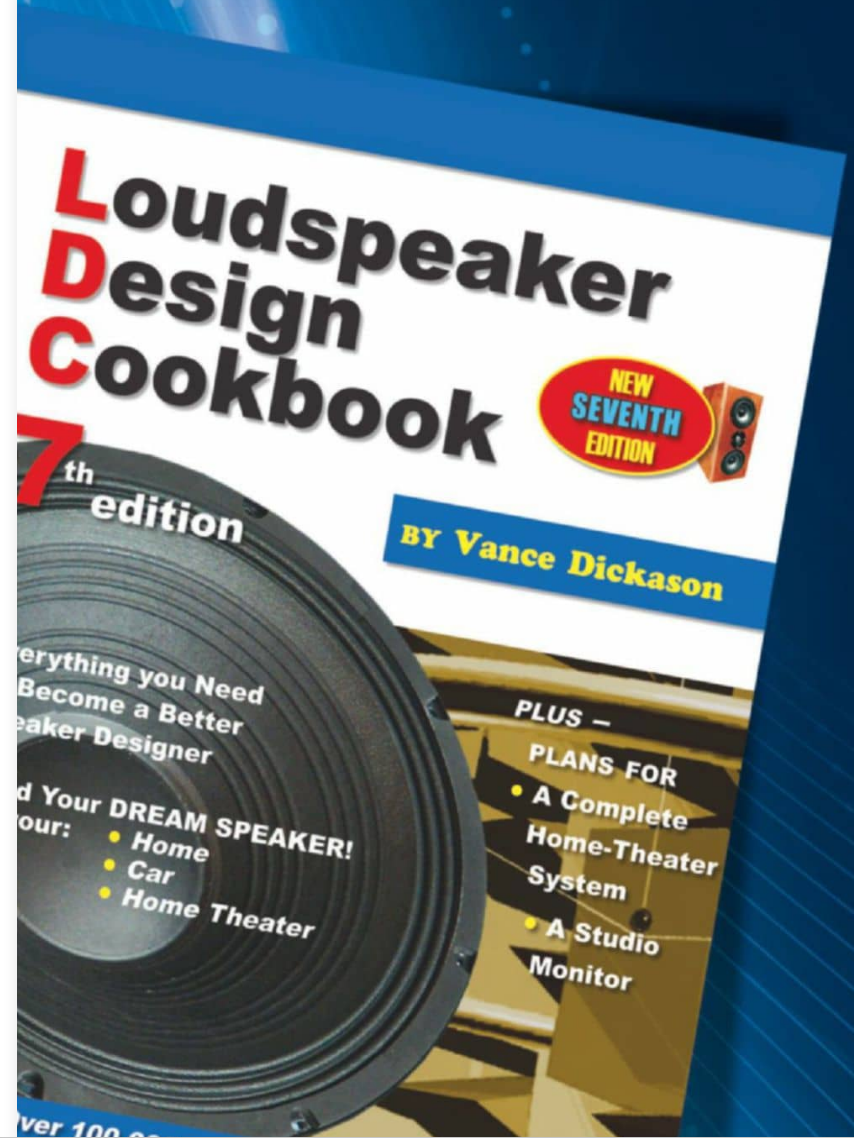

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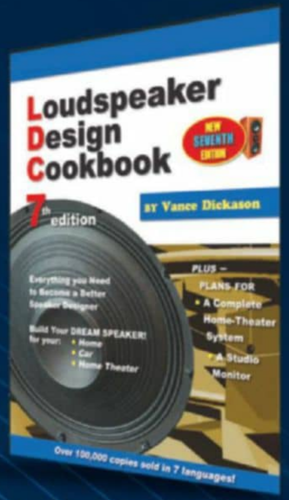

equipment. It quickly disconnects the output relay when the preamplifier is powered off or when a sudden power loss happens.

Using the MUTE switch, the output can be manually disconnected. The circuit also protects against a malfunction that could produce high DC voltage at the output of the amplifier, protecting the expensive headphones.

When the power is applied to the circuit, the components Z2, R76, and Q4 detect the presence of the DC voltage and when this voltage goes higher than about 25VDC, the transistor Q4 is set to the On state. This also activates transistor Q3 which short circuit the diode D9 to the ground deactivating the transistor Q5.

The Zener diode Z1 generates a 12VDC constant voltage, which is used for the output relay's activation in conjunction with the capacitor C1's voltage, which is increased slowly by R78. When it reaches about 12.6VDC, the transistor Q9 is activated and turns on the transistor Q6, which turns on the relay K5. The LED_{out} is connected in parallel with the output relay through the current limited resistors R98 and R99. When the LED illuminates, it indicates that the output is normally connected.

D15 with R92 are used to activate transistor Q5 with an external voltage coming from the manual MUTE switch. In this case, Q5 is turned on and deactivates the relay K5 through transistors Q9 and Q6.

Any large DC offset at the output is sensed by the circuit around transistor Q7 and Q8 with diodes D7, D8, D12, D13, D14, and D20. The components R22, C34, and C44 for the Left Channel and R85, C33, and C43 for the Right Channel create a low-pass filter at a very low frequency to avoid activation of the protection circuit by strong bass frequencies of the output signal.

The Construction of the PhL-100

I used the Design Spark PCB software to design the PCB. The outcome was a 200mm × 130mm double-sided PCB with ground planes on the top and bottom layers. One PCB is used for both Left and Right channels.

Figure 7 shows the PCB's complete guide assembly. The Parts List shows what is required to build the preamplifier. I used high-quality parts for the preamplifier's construction like 0.1% resistors and polypropylene capacitors where needed.

For the assembly of the PCB, I started by placing and soldering all the components of the Power Supply section. I connected the power to the circuit and I checked that the voltages at pin 8 and pin 4 of all the op-amps of the Line and Headphone amplifier were about 0.1VDC less than the input voltage. The voltages at the op-amps pins of the Phono section should be about 2VDC less. I did this to be sure that all the SMD op-amps will find the correct voltage when they will be soldered on the PCB. This eliminates the risk to destroy the expensive op-amps if something goes wrong with the power supply.

Then I removed the power and I soldered all the SMD op-amps (8-lead plastic SOT-23 package) with my standard soldering iron using a very thin tip and a 0.38mm solder wire. A desktop magnifier with a LED light helped me do the job correctly. The assembly of the other through-hole components is very easy and it should be done from the smaller to the larger components.

Resistors R22 and R85 should not be soldered to the PCB as they will be used later to check the protection circuit. I used a metal enclosure with an aluminum front panel to house the components of the preamplifier. The enclosure had the following external dimensions: Width= 270mm, Depth=200mm, and Height=45mm.

Figure 8 shows the internal wiring of the PhL-100. To eliminate any possible noise at the output when switching between the three inputs, I inserted two extra MUTE positions between them. For example, to switch from CD to the AUX input, the switch goes first to the MUTE position which disconnects the output and then to the AUX position where the output is connected again with a delay.

On the left side of the PhL-100's front panel (Photo 1), the red LED indicates the operation and the green LED indicates that the output is connected. Next, there is the Input Selection switch and in the middle is the headphone connector. Between them, one 3mm red LED indicates that the muting circuit is active. I placed the volume potentiometer on the right side of the panel.

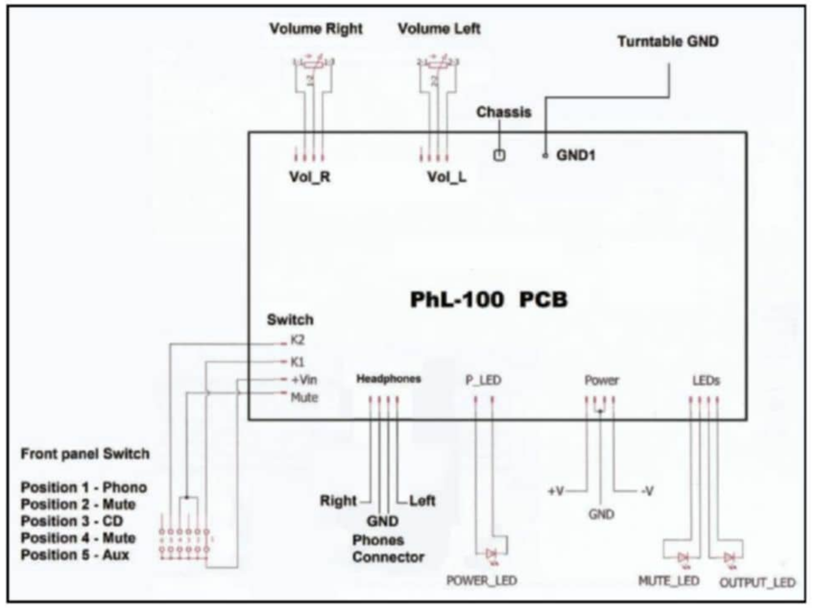

Figure 8: This diagram shows the internal wiring of the PhL-100.

The text on the front plate was created using the "Front Panel Designer" program (see Resources). The file was printed on a transparent self-adhesive sheet with my ink-jet printer.

The back panel of the preamplifier is shown in **Photo 2**. On the right side, there is a 5-pin DIN connector for the supply of the preamplifier. On the left side, there are eight gold-plated RCA female connectors for the inputs and the outputs of the preamplifier. They are placed with the same order as that of the PCB giving a very neat internal wiring as shown in **Photo 3**.

Testing the PhL-100

After the construction was complete, I checked all the power supply voltages for both channels. I checked also that the servo circuits were operating within their limits. This means that the voltage at pins 1 and 7 of IC3 and IC4 should be between -2.5VDC to +2.5VDC or so. In my prototype, the voltage ranged from -0.3VDC to -1.5VDC.

The protection circuit should be tested and ensured that it operates correctly when a DC fault appears at the output. As I mentioned, R22 and R85 were not soldered on the PCB. To do the testing, the pins of R22 and R85 that are close to the K5 relay are left free and the other pins are soldered to the PCB. This is necessary because an external voltage will be connected to the free pins and I didn't want to put the outputs of the op-amp's IC6, IC7, IC9, and IC10 at risk.

I connected an external positive voltage of +3VDC directly to the free pin of R22 with the ground of the +3VDC connected to the ground of the PhL-100. This should deactivate the output relay disconnecting the output. I repeat the same with a negative voltage of -3VDC.

The same procedure is performed for the other channel, with the external voltage now connected to the free pin of R85. After the successful completion of the test, the free pins of R22 and R85 are soldered to the PCB.

One final thing to notice is that both preamplifier and headphones outputs are active simultaneously. This could be inconvenient if someone wants to only listen with the headphones. I don't have that problem since I can mute the DSP digital crossover with my system.

One solution can be a high-quality switch that would disconnect the preamplifier's line output signal. Another option would be a switched headphone connector (e.g., the Neutrik NYS216G), which would switch off the line out when the headphones are connected.

Measurements of the PhL-100

I measured the accuracy of the preamplifier's RIAA curve with an accurate inverse RIAA network, which was constructed with high-accuracy components. For more details about this network see my article "The MC100 A High-Quality Moving

Photo 2: The back panel has a 5-pin DIN connector and eight gold-plated RCA femail connectors.

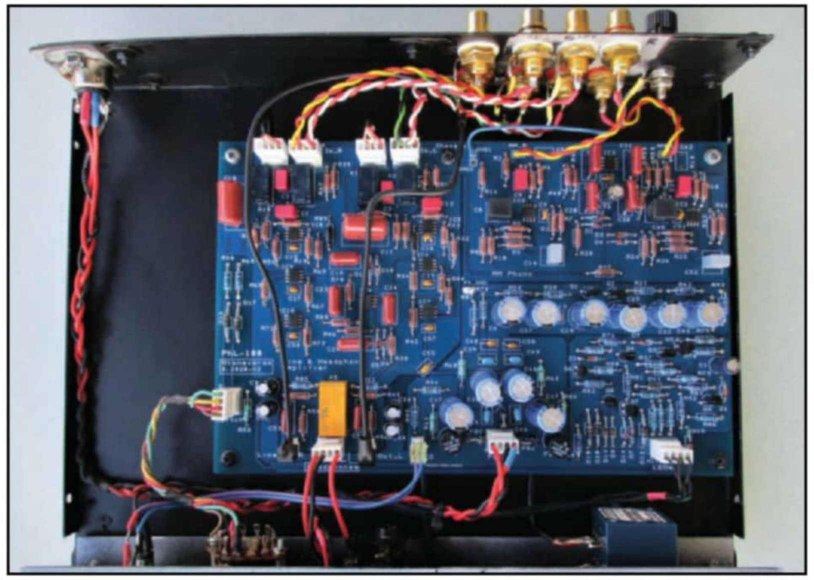

Photo 3: The eight gold-plated RCA female connectors for the inputs and the outputs of the preamplifier are placed with the same order as that of the PCB.

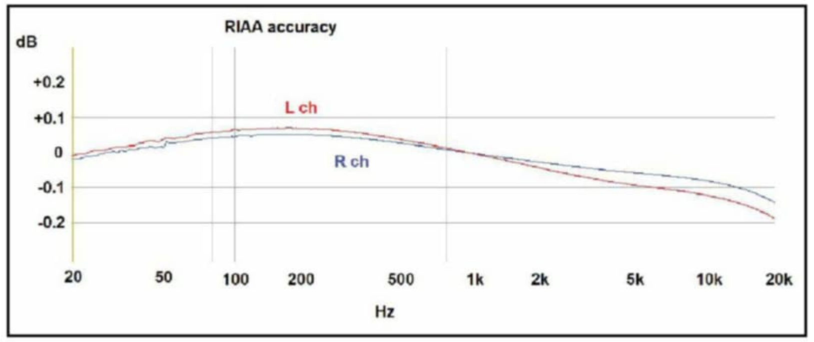

Figure 9: The accuracy of the RIAA response of the PhL-100 was tested.

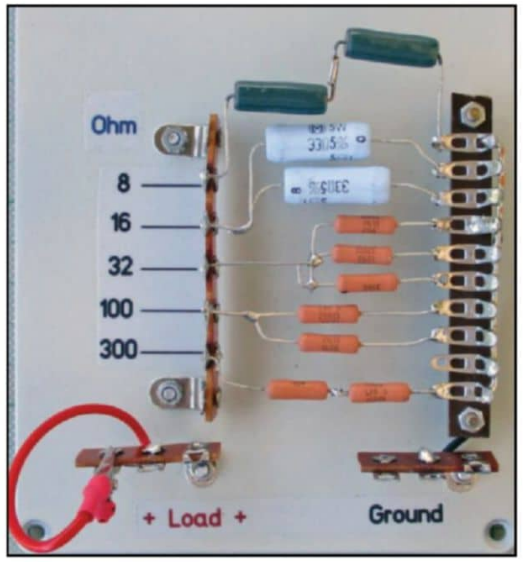

Photo 4: I built a dummy load to test the PhL-100.

Coil RIAA Preamplifier" (*audioXpress*, March 2014) **Figure 9** shows the preamplifier's accuracy with this inverse RIAA network connected at its input. The deviation from the ideal RIAA curve is about ±0.1dB from 20Hz to 10kHz. Of course, this includes the error of the inverse RIAA network and the preamplifier.

The input impedance of the line preamplifier at 1kHz ranges from 30kΩ to 50kΩ, depending on the position of the Volume potentiometer. The maximum gain of the line preamplifier is 21.6dB. The -3dB points of the frequency response are at 2Hz and150 kHz.

The impedance of the headphones output is 2Ω and the maximum output voltage and power depends on the connected load. I built a dummy load as shown in **Photo 4** and I measured the following:

9.7Vrms	300Ω	0.3 W
9.5Vrms	100Ω	0.9 W
5.9Vrms	32Ω	1.1W
3 Vrms	16Ω	0.56 W
1.3Vrms	8Ω	0.21W

The impedance of the line output is 100Ω and the maximum voltage is 10Vrms at 2kΩ. I measured the distortion at 1kHz in all the described loads from 8Ω to 300Ω, keeping the level constant at 1Vrms. I never measured distortion above 0.001%, which unfortunately is the limit of my distortion analyzer. I am sure that the real distortion is much lower.

A very important issue that affects the quality of the stereo reproduction is the imbalance between the two channels. This is mainly due to the volume potentiometer since the gain of the preamplifier is set with high accuracy resistors. I measured this by moving the potentiometer to various positions and I noticed that the Left Channel was always greater in level than the Right Channel, from 0.15dB to a maximum of 0.3dB. I decided to increase the gain of the Right Channel by 0.22dB by soldering a resistor of 15.6kΩ in parallel to R56 (470Ω). The final result was that the imbalance between the two channels drop to ±0.1dB, at least for the positions around the middle, which are the most frequently used in listening.

The PhL-100 uses a common PCB for both channels and I took every precaution during the design and construction to avoid any interference from one channel to another. Testing the cross-talk between the channels proved that the result was successful. The test was performed at 1kHz, 10kHz, and 20kHz with the output voltage of one channel at 10Vpeak and the input of the other channel open. The results were -81dB at 1kHz, -69dB at 10kHz, and -64dB at 20kHz.

Conclusion

The PhL-100 is a versatile and easy-to-construct high-quality device. It has an input for a moving magnet cartridge and additionally two line inputs. It can easily drive every power amplifier and has a lot of power to drive almost any headphone. It is very, very quiet and offers excellent transparency. If you need such a device, I am sure you will enjoy building and listening to this one.

Author's Note: I have PCBs available for the construction of the PhL-100 device. If you are interested, email me at gntanavaras@gmail.com

About the Author

George Ntanavaras graduated from the National Technical University in Athens, Greece, in 1986 with a degree in Electronic Engineering. He currently works in the Development Department for a Greek electronics company. He is interested in the design of preamplifiers, active crossovers, power amplifiers, and most loudspeakers. He also enjoys listening to classical music.

Resources

DesignSpark, www.rs-online.com/designspark/pcb-software

Front Panel Designer, www.schaeffer-ag.de

G. Ntanavaras, "The MC100 A High-Quality Moving Coil RIAA Preamplifier," *audioXpress*, March 2014
https://audioxpress.com/article/the-mc100-a-high-quality-moving-coil-riaa-preamplifier

G. Ntanavaras, "A High-Quality, High-Power Headphone Amplifier," *audioXpress*, June 2013
https://audioxpress.com/article/Build-A-High-Quality-High-Power-Headphone-Amplifier

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You Can DIY!

Build a Current Feedback Headphone Amplifier

Photo 1: The completed current feedback headphone amplifier project offers low noise and low distortion.

By
Brian Bylenok

Inspired by notable current feedback amplifier projects previously published by *audioXpress*, the author proposes a refined headphone amplifier design with very low noise and very low distortion that resulted from methodical experimentation.

I have been interested in current feedback amplifiers since *audioxpress* first published the "Willow" article by Robert Nance Dee in December 2011. His follow-up article, "Willow Revisited" renewed my interest. But it was the March 2021 Audio Scope article "Very Low Distortion, Greater Than 400mA Output Composite Amplifier Headphone Driver" by Michael Steffes that really kicked me into gear. I've built a half dozen different DIY headphone amplifiers over the years, finally settling about a year ago on Wayne Colburn's "Whammy" design. Since Steffes' design also incorporated the OPA1612, my all-time favorite monolithic op-amp, I figured it was time to take up the challenge and see if a "chip" headphone amplifier (HPA) could compete with a well-regarded Class-A HPA design. **Photo 1** shows the end result of my labor.

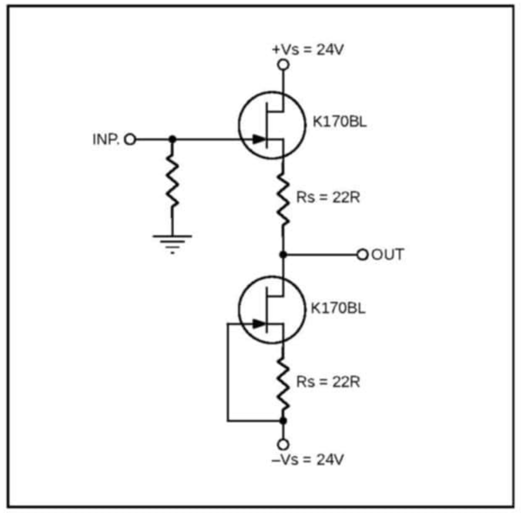


Figure 1: Erno Borbley's circuit concept is similar to the one implemented in the Pass B1K preamp and in the Colburn BA2018 line amplifier.

The Design

This being a surface-mount device (SMD) project, I did a layout in KiCAD and sent off for some prototype circuit boards. I'm a believer in maintaining absolute polarity in the signal chain, at least with loudspeaker systems (more on that later). So I first tried a non-inverting version of Steffes' circuit design. That did not turn out to be a very satisfactory solution. Although the resulting sound offered a great amount of detail and solid well-controlled bass response, listening fatigue quickly set in. So I decided to work with his original inverting design.

But there was something else that bothered me besides the signal inversion. It was the circuit's inherently low input impedance. Although this is typically not a problem with most modern electronics working into a fixed load, introducing a passive volume control at the input of the circuit gave me varying results in tone and detail. The sound character changed depending on the potentiometer value and setting. So I decided to try adding an impedance matching unity-gain input buffer. I first did some experimentation with an OPA1612 as a unity gain buffer, both in an inverting and a non-inverting configuration, only to find the op-amp buffer detracted from the overall experience. I gave that up and eventually decided to explore a simple JFET input buffer.

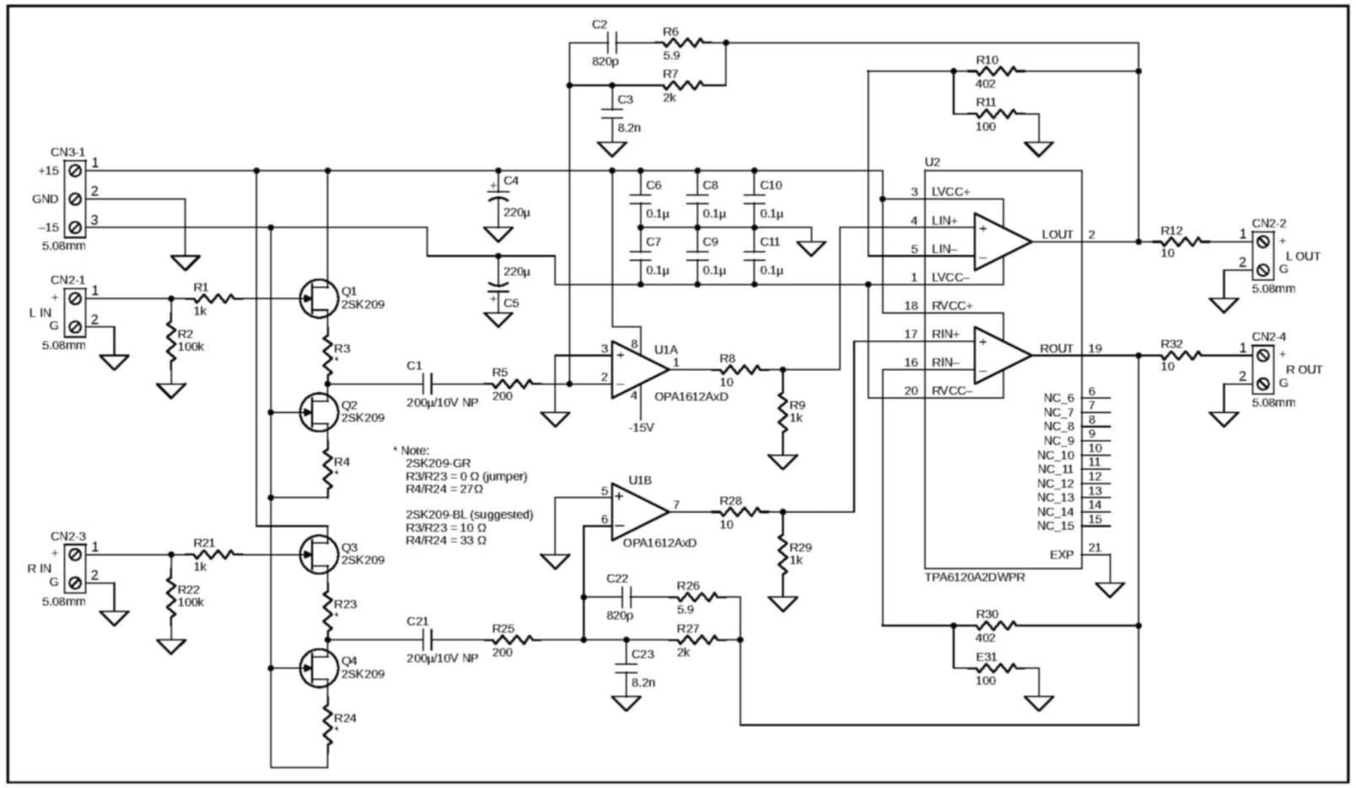


Figure 2: My final schematic details the design, with the JFETs and the two ICs as the only SMD components. Everything else is a thru-hole device to simplify DIY construction.

Prototypes with a JFET Input Buffer

For that, I turned to another technical article, Erno Borbely's "JFETs: The New Frontier, Part 2." Through my participation in the DIYaudio forums, I have studied and built some of the designs that Nelson Pass and Wayne Colburn of Pass Labs have created and donated to the DIY community.

Among them are variations of the circuit Borbely shows in Figure 15C of his article (**Figure 1**). His circuit concept is similar to the one implemented in the Pass B1K preamp and in the Colburn BA2018 line amplifier. I modeled a buffer based on those designs in LTSpice and built several prototypes. The first prototype used matched Fairchild J113 JFETs with source resistors selected to operate them in the

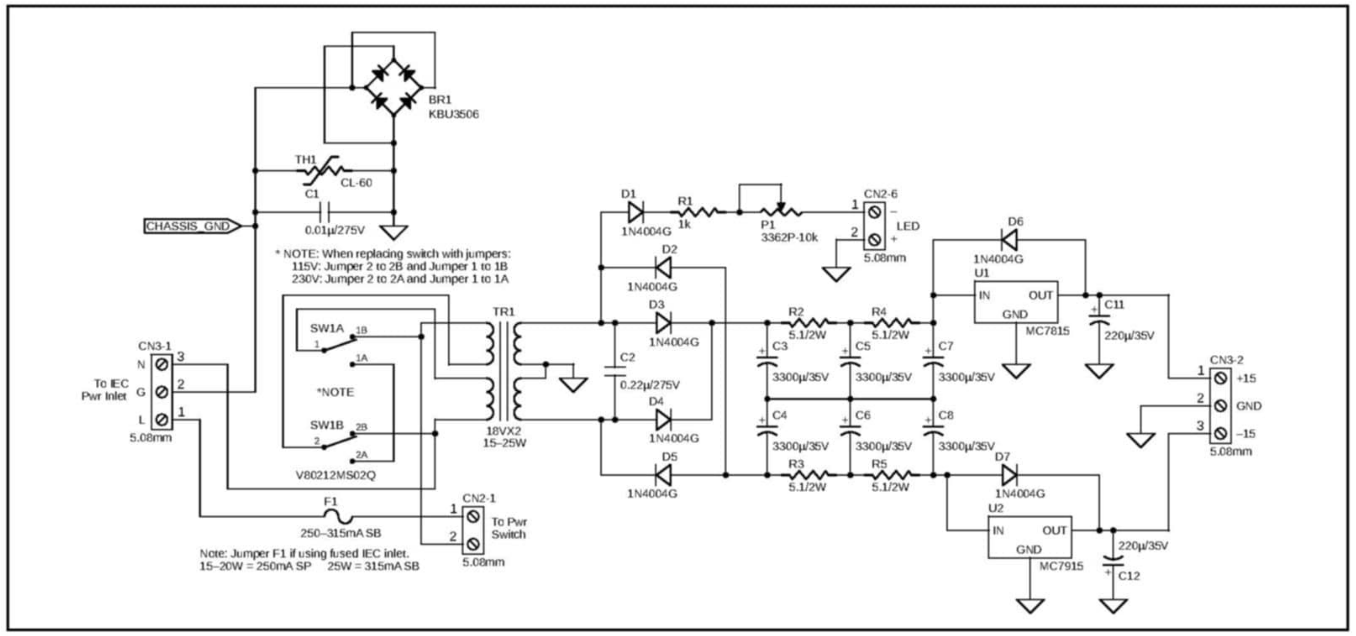
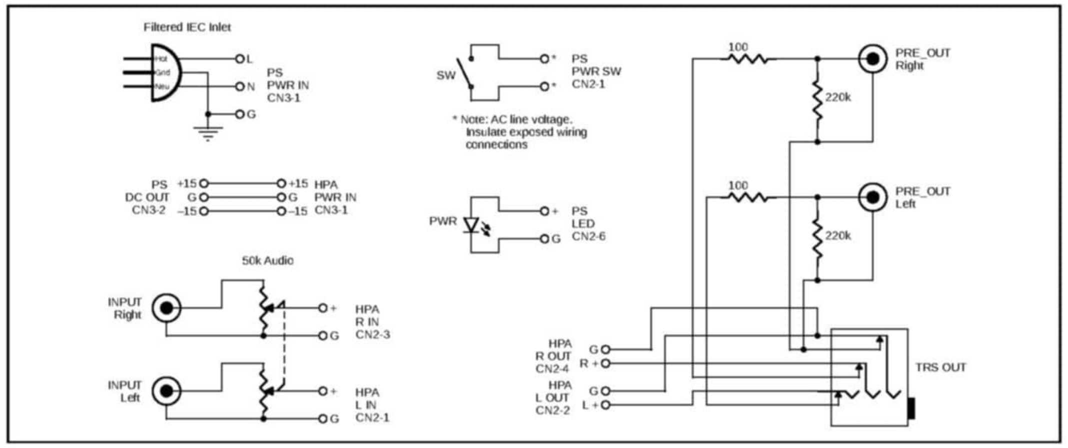


Figure 3: For the power supply, I used a 20W, ±15VDC regulated linear supply.

Figure 4: A switching headphone jack with connections to some RCA outputs with local loading resistors provide a separate preamp output.



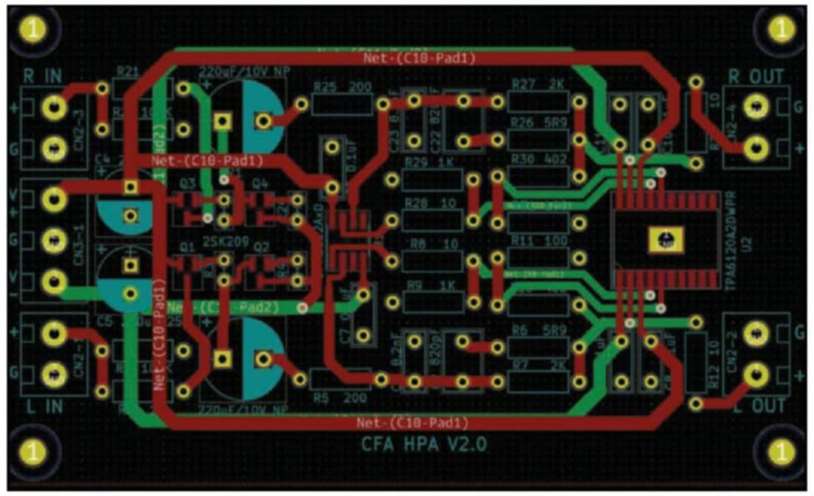


Figure 5: The main amp PCB layout was done with both a top and a bottom layer ground plane, along with power supply bypass caps adjacent to each of the IC power supply pins.

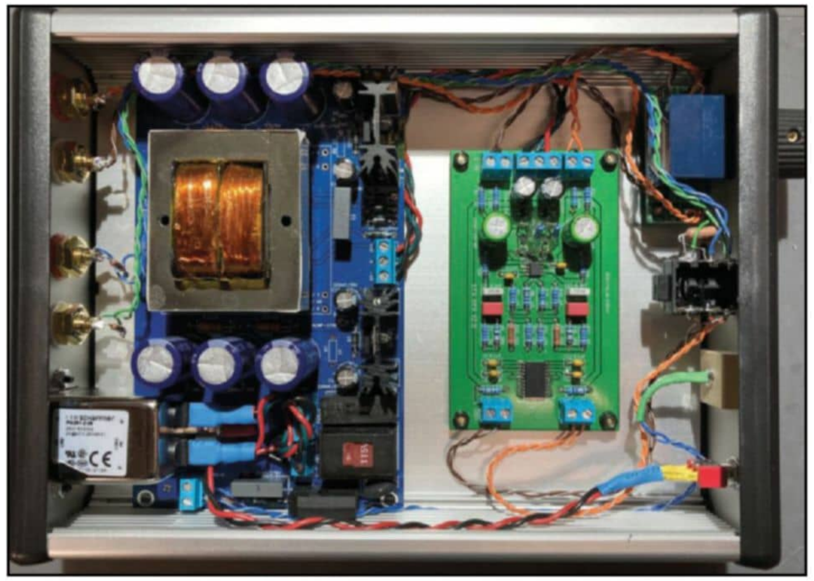


Photo 2: This top view photo includes the final version of the circuit board.

"sweet spot." The second prototype used the latest SMD implementation of the Toshiba 2SK209 JFETs. The Toshiba design using -GR grade devices and 27Ω source resistors was the clear winner. Due to lack of a suitable test fixture to accommodate the tiny SMD JFETS, I didn't do any Toshiba parts matching. Unless the JFETs are extremely well matched, you are likely to find some DC on the output of the buffer. I found 1mV of DC on one channel and about 7mV on the other.

To block that, I added a Nichicon MUSE 220µf bi-polar electrolytic capacitor in series with the buffer output. I also added 10Ω series resistors to the amplifier outputs to help stabilize the overall circuit.

This is less resistance than that shown in the Texas Instruments (TI) application note, but I have not experienced any problems so far. The final schematic is shown in **Figure 2**. The JFETs and the two ICs are the only SMD components. Everything else is a thru-hole device to simplify DIY construction. Even though positions for resistors R3/R4 and R23/R24 are typically not needed for the -GR devices, I added them along with some suggested values for the -BL grade devices. This layout will also provide for experimentation with source current of the -GR devices as well. Unfortunately, the pin out for the SMD version of the J113 is different but the low cost of the 2SK209 devices tends to make that moot.

The Power Supply

There's nothing special about the power supply and any quiet, well-regulated design of the proper voltages and current capability should work fine. I used a 20W, ±15VDC regulated linear supply (**Figure 3**). It's similar to Colburn's "Whammy" power supply design, running in "naked regulator" mode without the LED references that would raise the supply to more than ±17V. A 50kΩ audio taper Alps RK27-series potentiometer and a filtered IEC inlet round out some of the other choices.

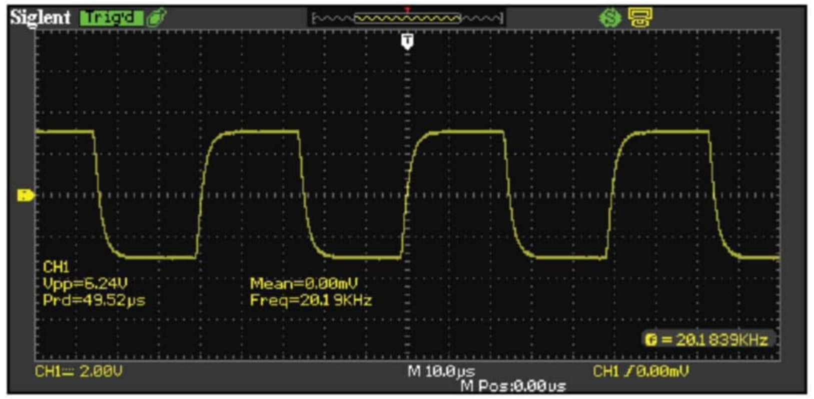


Figure 6: This test measurement shows the current feedback amplifier at 20kHz at 1W output into 39Ω

A switching headphone jack with connections to some RCA outputs with local loading resistors provide a separate preamp output (**Figure 4**). The main amp PCB layout was done with both a top and a bottom layer ground plane, along with power supply bypass caps adjacent to each of the IC power supply pins. I changed the originally specified TPA6120A2 RGY package to the larger DWPR package to simplify DIY construction (**Figure 5**). I also added a plated 2mm thru-hole centered under the TPA6120A2 heatsink connection pad. That was so the board would not require solder paste with a DIY oven, a heat gun, or a frying pan melting process and could be assembled with a regular soldering iron.

SMD construction ideally requires a good optical magnifier, a very fine, grounded tip temperature-controlled soldering iron and a grounded assembly mat with wrist strap. I was reluctant at first to try SMDs but now that I'm used to working with them I'm glad I did. A lot of good parts are only available in that form factor. One trick I have found with SMDs is to use a flux pen to apply extra flux to the PCB device lead pads and also under the amp chip at the heatsink pad location. The really tough devices to mount are the JFETs since they are as small as grains of rice and tend to just blow away if you are not careful. I made up a slender wooden stick with some duct tape mounted to the tip sticky side out for dealing with those pesky little critters. **Photo 2** shows a top view of the finished project with the cover off.

Final testing of the prototype included checking with an oscilloscope for possible high-frequency oscillation and with a meter for DC on the outputs. Neither problems were found, and the circuit appears very stable.

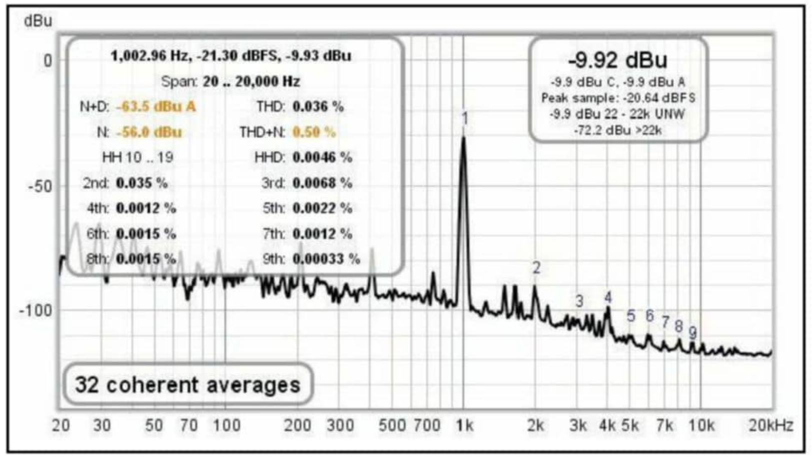


Figure 7: This graph shows the current feedback amplifier's THD measured at -10dBu ouptut.

Sound Tests

How does it sound? I can only say that it more than lives up to Steffes' design model. It performs equally well with all of the headphones that I have in my collection. The music is set against an inky background of dead silence. This headphone amplifier has a very smooth, non-fatiguing sound, with no hint of harshness on transients, yet the amount of musical detail revealed has me hearing new things in material I thought I knew very well.

There really is something to appreciate about a very low noise and very high slew rate circuit that also appears to exhibit a very low level of total harmonic distortion (THD) and transient intermodulation (TIM) distortion. As far as the signal polarity inversion, it turned out to be a non-issue. I have one pair of headphones that has balanced 2.5mm wiring connections at each ear-cup.

I built a 4PDT polarity reversing switch box and did some listening evaluations. I paid extra attention to kick drum transients and other low-frequency information that logically seemed like it might be affected. **Figure 6** shows the current feedback amplifier at 20kHz at 1W output into 39Ω. **Figure 7** shows the current feedback amplifier's THD measured at -10dBu output.

I found that I could hear absolutely no difference in any aspects of the music when switching polarity. This headphone amplifier offers truly remarkable sound quality and more than holds its own against other well-regarded designs.

About the Author

Brian Bylenok's first electronics project was a cat whisker detector AM radio he built in the late 1950s. Ham radio and Heathkit tube audio amplifiers soon followed. Before retirement, he worked in the audio and technology systems industry for more than 40 years. His professional audio experience includes product and systems design in consumer, commercial, MI, and pro sound. Besides DIY audio, his hobbies include piano tuning and using his home recording studio. He and his life partner of 30 years live in a self-designed, mostly self-built home in a remote part of the Sonoran desert outside of Tucson, AZ.

Project Files

To download additional material and files that include Parts Lists for the Main PCB, Power Supply and Chassis as well as the PCB fabrication Gerber and drill files, visit: http://audioxpress.com/page/audioXpress-Supplementary-Material.html

Resources

"B1K Korg Triode," DIYaudio Forum, www.diyaudio.com/forums/pass-labs/313612-b1-korg-triode.html

E. Borbely, "JFETs: The New Frontier: Part 2," reprinted by *audioXpress*, December 2015, https://audioxpress.com/article/JFETs-The-New-Frontier-Part-2

R. Nance Dee, "Willow," *audioXpress*, December 2011, https://audioxpress.com/article/The-Willow-Pre-amp-A-high-slew-rate-JFET-amplifier

R. Nance Dee, "Willow Revisited," *audioXpress*, February 2016, https://audioxpress.com/article/willow-revisited-a-design-celebrating-the-enthusiasm-and-the-creativity-of-the-builder

N. Pass, "The Sweet Spot," Pass Labs, https://www.passlabs.com/technical_article/the-sweet-spot

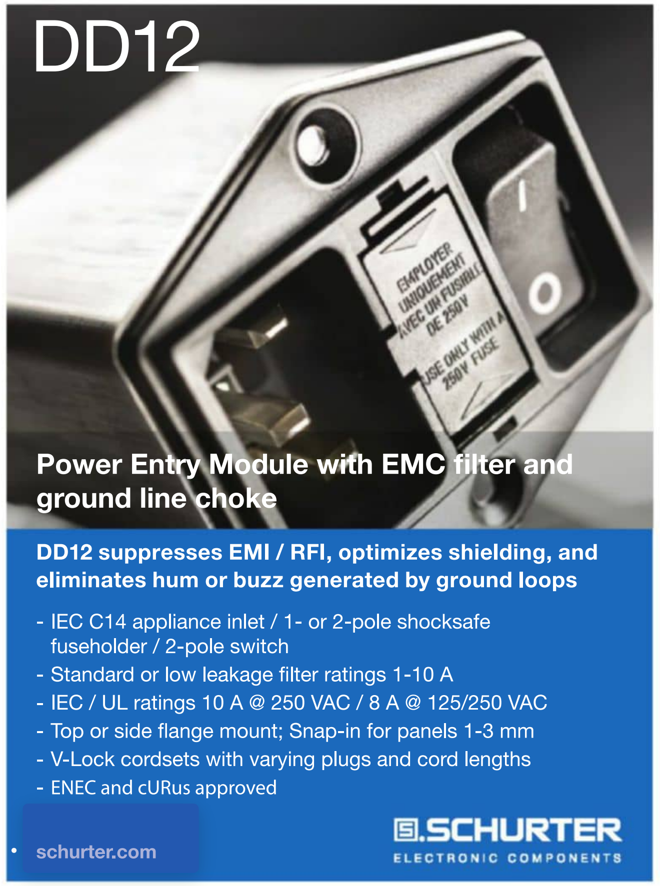
M. Steffes "Very Low Distortion, Greater Than 400mA Output Composite Amplifier Headphone Driver," *audioXpress*, March 2021.

"TPA6120A2 High Fidelity Headphone Amplifier," Texas Instruments, revised February 2015, Simplified Schematic 4, www.ti.com/lit/gpn/tpa6120a2

"Waynes BA 2018 Linestage," DIYaudio Forum, https://www.diyaudio.com/forums/pass-labs/329240-waynes-ba-2018-linestage.html

"Whammy Pass DIY headphone Amp Guide," DIYaudio Forum, www.diyaudio.com/forums/pass-labs/317803-whammy-pass-diy-headphone-amp-guide.html

Editor's Note: All audioXpress articles from 2001 to present can be found on the aX Cache, a USB drive available from www.cc-webshop.com.



Tube Power Amps Have Classes

Richard Honeycutt discusses the distinctive features of various amplifier circuits and what characterizes them as Class A, Class B, Class C, Class D, Class AB_1, and Class AB_2.

By
Richard Honeycutt

The history of hollow-state power amplifiers has been one of challenges and innovations. For a fuller discussion of these amplifiers' operation, see the May and June, 2012 Hollow-State Electronics articles in *audioXpress*.

The simplest hollow-state power amplifier is a single-ended Class A circuit such as the one used in the Fender Champ guitar amplifier (**Figure 1**). Many DIYers built similar amplifiers for their home music systems. Note that the sine-wave input signal is loosely matched by the higher-amplitude sine-wave output signal. Class A amplification is defined by the active device (tube, in this case) conducting throughout the entire cycle of the signal.

Class A amplifiers have limited efficiency—25% maximum. Thus even using high-power triodes or pentodes, single-ended Class A amplifiers are limited in maximum power output because of the finite maximum plate power dissipation of tubes. There are two ways to increase the power output of Class A amplifiers: parallel and push-pull topologies (**Figure 2**). In the classic Western Electric Model 92 power amplifier used in many of the best cinemas in years past, the 300A power triodes were connected in parallel (**Figure 3**). I should also mention that the principles discussed in this article apply equally to triodes and triode-connected pentodes.

The push-pull topology uses two output tubes, each amplifying one half-cycle of the signal. An amplifier that only conducts for the positive or negative half-cycle—180°—is classified as Class B. Class B push-pull design is almost never used, since a "crossover notch" is created in the signal as the tubes switch from nonconducting to conducting or vice-versa. This notch is heard as an especially annoying form of distortion. It can be prevented by biasing both tubes so that they can never both be in a nonconducting state at the same time. This biasing produces what is called Class AB operation. This class is divided according to the bias and signal levels into Class AB_1, which has a more negatively-biased control

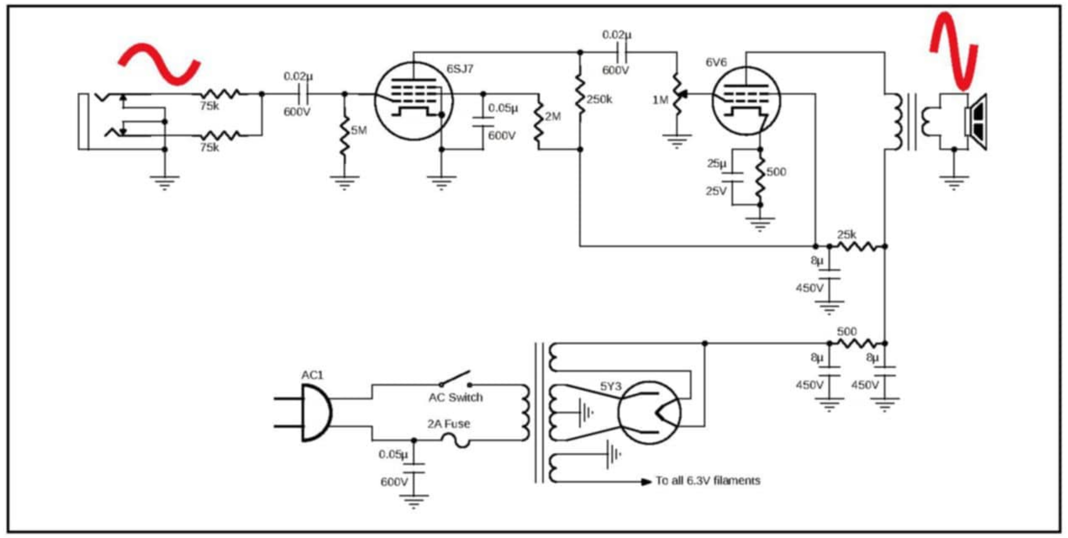


Figure 1: This Fender Champ single-ended Class A amplifier is similar to many low-power DIY hollow-state amplifiers of the past.

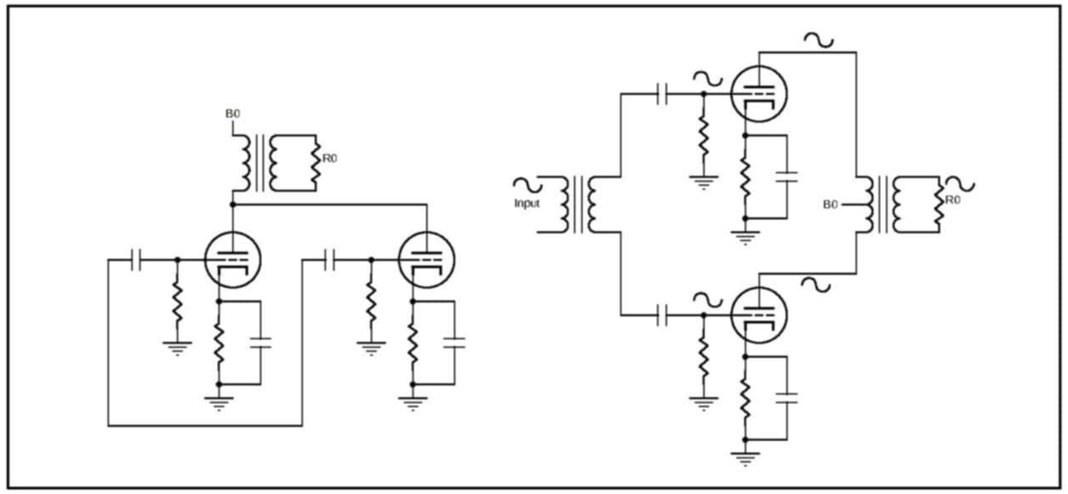


Figure 2: Class A amplifiers may be made to produce higher output powers by the use of parallel or push-pull topologies.

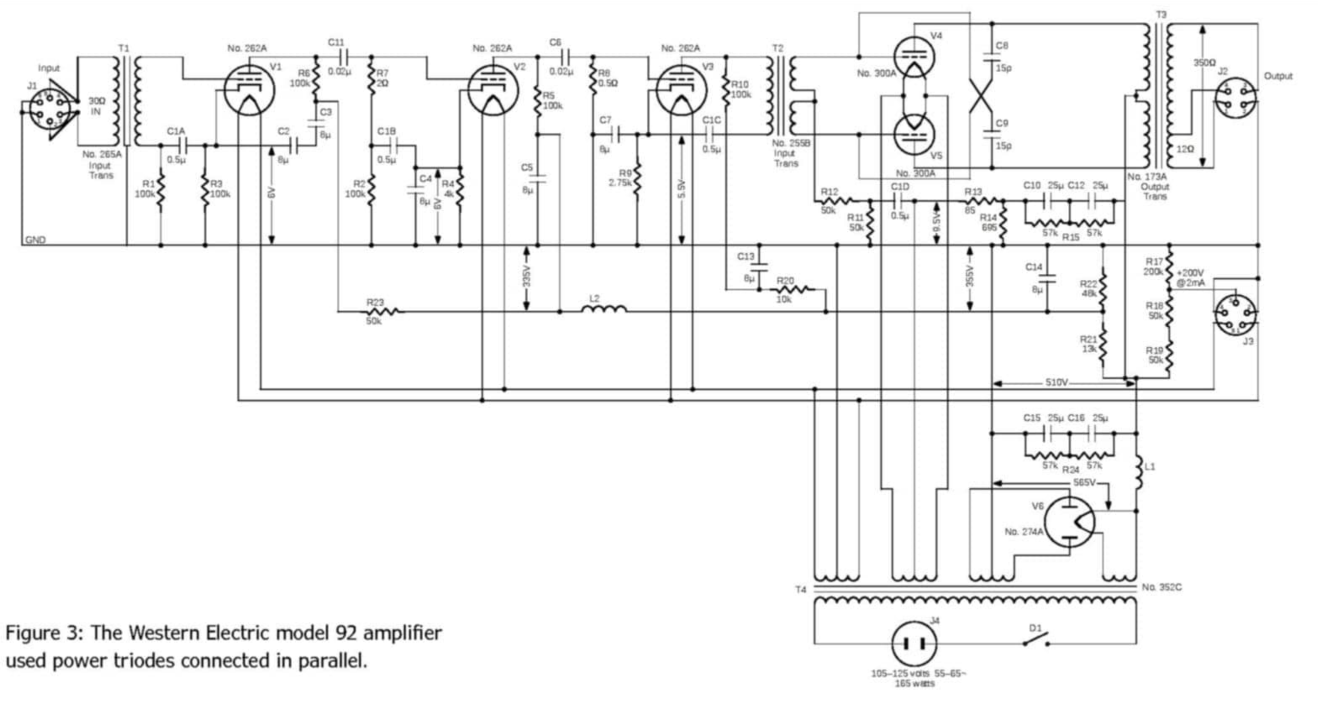


Figure 3: The Western Electric model 92 amplifier used power triodes connected in parallel.

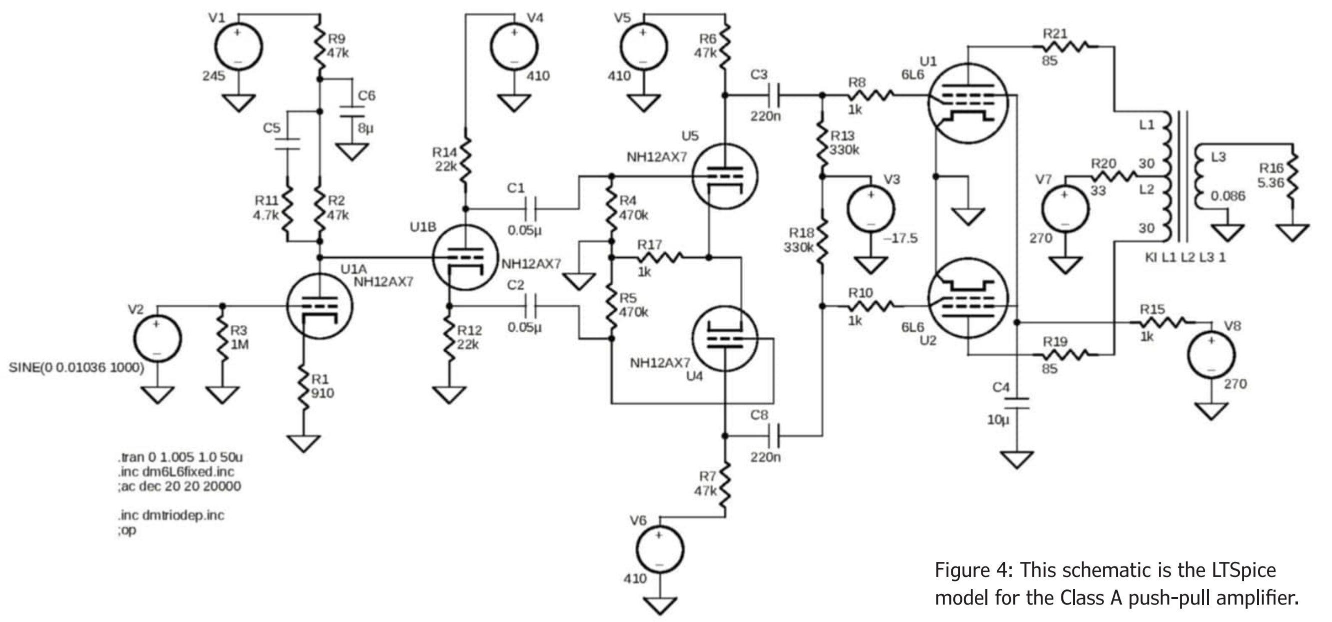


Figure 4: This schematic is the LTSpice model for the Class A push-pull amplifier.

Table 1: RCA suggested the operating conditions listed here for a 6L6GC Class A push-pull design.

Class A Push-Pull				
Operating Conditions	**Fixed**	**Bias**	**Cathode**	**Bias**
Plate Supply Voltage	250V	270V	250V	270V
Grid No.2 Supply Voltage	250V	270V	250V	270V
Grid No. 1 Voltage	-16V	-17V	–	–
Cathode Resistor	–	–	124Ω	124Ω
Peak AF Grid No.1 to Grid No.1 Voltage	32V	35V	35.6V	28.2V
Zero Signal Plate Current	120mA	134mA	120mA	134mA
Maximum Signal Plate Current	140mA	155mA	130mA	145mA
Zero Signal Grid No.2 Current	10mA	11mA	10mA	11mA
Maximum Signal Grid No. 2 Current	16mA	17mA	15mA	17mA
Plate Resistnace (appoximate, per tube)	24500Ω	23500Ω	–	–
Transconductance (per tube)	5500μmhos	5700μmhos	–	–
Effective Load Resistance (plate to plate)	5000Ω	5000Ω	5000Ω	5000Ω
Total Harmonic Distortion (THD)	2%	2%	2%	2%
Maximum Signal Power Output	14.5W	17.5W	13.8W	18.5W

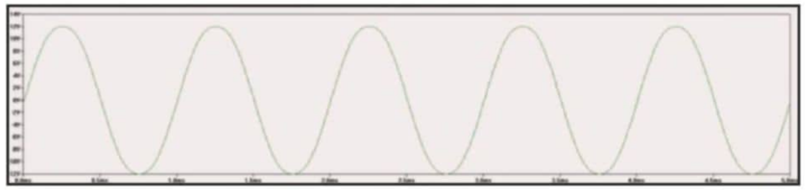


Figure 5: The output wave of the Class A push-pull circuit looks almost perfect.

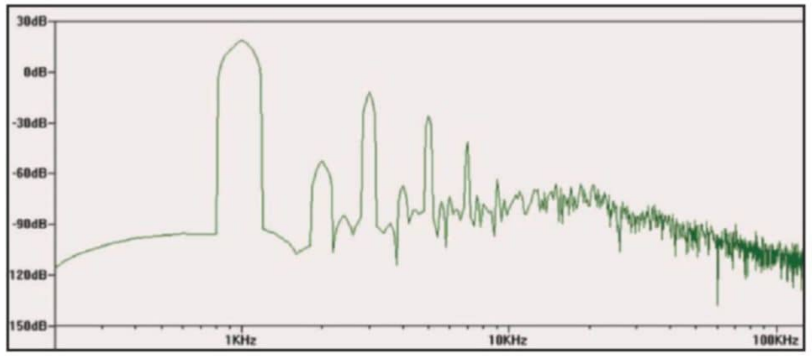


Figure 6: The FFT of the Class A output wave shows a preponderance of odd harmonics.

About the Author

Dr. Richard Honeycutt bought a *Popular Electronics* magazine from the rack at a bus station in 1960. It contained an article on building a transistorized audio amplifier, which captured his interest. He followed up by subscribing to several electronics magazines. To avoid the cost of parts, he built up a "junk box" by disassembling trade-in TVs from his uncle's furniture store. His dad bought him the Van Valkenburg, Neville, and Nooger Basic Electronics book set, so he learned about tube circuits as well as transistors. He started repairing electronic devices about 1965, earning his First-Class Commercial FCC license in 1969. He has worked part-time in broadcast radio engineering and audio electronics repair since 1968, in addition to full-time work in acoustical and audio system design, plus a 20-year stint teaching electronics at the college level.

grid, and Class AB_2, whose control grid is not as negatively biased, and which often is driven by a larger-amplitude signal. Control-grid current is always zero in a Class AB_1 amplifier, but sometimes there is nonzero control-grid current in a Class AB_2 amp.

There are two other amplification classes: Class C and Class D. Class C amplifiers conduct for less than 180° and are used mainly for RF amplification of one or a small range of frequencies. In these applications, the remainder of the waveform is "reconstructed" by a resonant circuit. Thus the output amplitude is constant. Class C amplifiers have traditionally been used as the final stage of FM radio transmitters. Class D amplifiers are switching designs—they produce pulses that are then converted into analog waveforms. Both Class C and Class D amplifiers have higher maximum efficiencies than Class B or AB, which are limited to 78.5%. Higher efficiency translates into a higher possible output power from a given pair of output tubes.

One issue that factors into the choice of Class A, AB_1, or AB_2 is the amount and frequency signature of the distortion. In order to compare these, I simulated hollow-state power-output circuits using LTSpice. I used the example circuits for the 6L6GC given in the *RCA Receiving Tube Manual*. The only negative feedback present is in the unbypassed cathode resistor of the input stage (**Figure 4**). **Table 1** shows the data given for Class A push-pull operation.

Class A Push-Pull

The specific design I chose is the one shown in the second column of Table 1, as can be seen from Figure 4. Note that no overall negative feedback is used. The output wave of the Class A circuit just before the onset of clipping is shown in **Figure 5**, and the Fast Fourier Transform (FFT) of that wave is shown in **Figure 6**. The almost picture-perfect sine wave actually has a total harmonic distortion (THD) of about 1.3%, mostly third harmonic.

Class AB_1 Push-Pull

Table 2 shows the RCA suggested operating conditions for a 6L6GC push-pull Class AB_1output stage. The circuit I modeled in LTSpice used the values shown in the third column of Table 2. As with the Class A circuit, no overall negative feedback was used.

Class AB_1 operation produces an output wave that looks very similar to that from the Class A amplifier, as you can see from **Figure 7**. The FFT is shown in **Figure 8**. Contrary to statements common in the literature, the Class AB_1 amp has slightly higher levels of lower-order harmonics—THD is about 1.4%—than does the Class A amplifier. This is true even for even-order harmonics, which are usually expected to be cancelled out by push-pull operation. But you can see that the odd-order harmonics predominate.

It is true that the maximum peak output voltage of the Class A stage is only about one-third of that produced by the Class AB_1 circuit, so the tubes were operating in different regions of their characteristic curves. Probably of more importance from an audio standpoint is the much (~40dB) higher levels of higher-order harmonics produced by the push-pull circuit.

Class AB_2 Push-Pull

Table 3 shows the operating conditions RCA suggested for a 6L6GC push-pull Class AB_2 design. My LTSpice model was based upon the values shown in the right-hand column of Table 3.

The output wave of the Class AB_2 circuit (**Figure 9**) is a bit more distorted than that of the Class AB_1 circuit: THD is 2.2%. But odd-order harmonics continue to predominate, and higher-order harmonics appear at much higher levels than for the Class A amp (**Figure 10**). In fact, little difference in distortion signature can be seem between the Class AB_1 and AB_2 waves, except that even-order harmonics above the sixth are actually lower in the Class AB_2 design.

As mentioned earlier, the biasing of Class A and AB push-pull amps precludes control-grid current. However, a small current does flow in the grids of output tubes biased for Class AB_2 operation, as you can see in **Figure 11**. The nearly triangular waveform indicates significant distortion in the grid current, compared to a sine wave.

Operating Conditions	Fixed	Bias	Cathode	Bias
Plate Supply Voltage	360V	450V	450V	360V
Grid No.2 Supply Voltage	270V	350V	400V	270V
Grid No.1 (Control Grid) Voltage	-22.5V	-30V	-37V	–
Cathode Resistor	–	–	–	248Ω
Peak AF Grid No.1 to Grid No.1 Voltage	45V	0.60V	70V	40.6V
Zero Signal Plate Current	88mA	95mA	116mA	88mA
Maximum Signal Plate Current	132mA	194mA	210mA	100mA
Zero Signal Grid No.2 Current	5mA	3.4mA	5.6mA	5mA
Maximum Signal Grid No.2 Current	15mA	19.2mA		22mA
Effective Load Resistance (plate to plate)	6600Ω	6000Ω	5500Ω	9000Ω
Total Harmonic Distortion (THD)	2%	1.5%	1.8%	4%
Maximum Signal Power Output	26.5W	50W	55W	24.5W

Table 2: These are the operating conditions in RCA's sample Class AB_1 design.

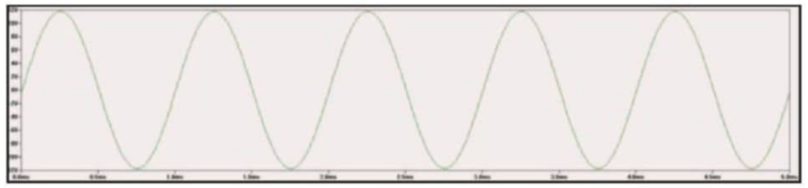


Figure 7: The output wave from the Class AB_1 amplifier is almost visually indistinguishable from that of the Class A amp.

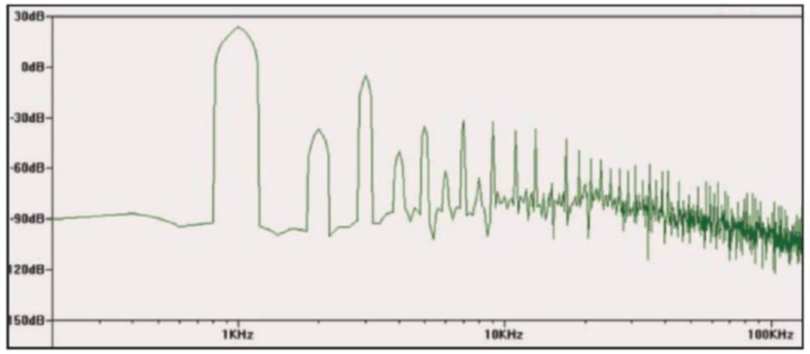


Figure 8: The FFT of the Class AB_1 output wave shows an increase in higher-order harmonics, compred to that of the Class A amplifier.

Operating Conditions	Fixed	Bias
Plate Voltage	360V	360V
Grid No.2 Voltage	225V	270V
Grid No.1 (Control Grid) Voltage	-18V	-22.5V
Peak AF Grid No.1 to Grid No.1 Voltage	52V	72V
Zero Signal Plate Current	78mA	88mA
Maximum Signal Plate Current	142mA	205mA
Zero Signal Grid No.2 Current	3.5mA	5mA
Maximum Signal Grid No.2 Current	11mA	16mA
Peak Grid Input Power	140mW	270mW
Total Harmonic Distortion (THD)	2%	2%
Maximum Signal Power Output	31W	47W

Table 3: RCA suggested these operating conditions for a 6L6GC Class AB_2 design.

Figure 9: Even though the circuit draws control-grid current, the Class AB$_2$ amplifier's output wave still looks as though it is relatively undistorted.

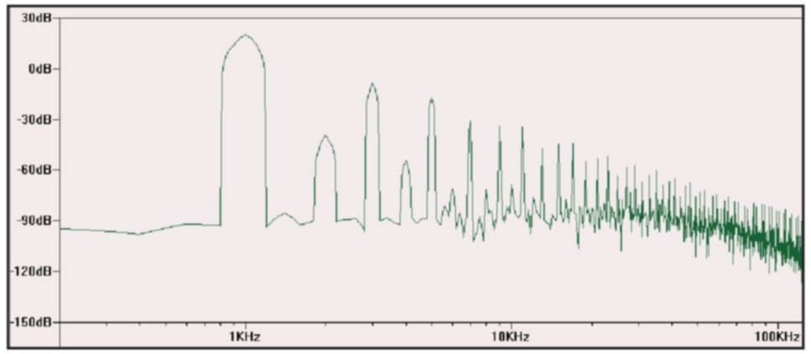


Figure 10: The FFT of the Class AB$_2$ output wave shows high upper-level harmonics.

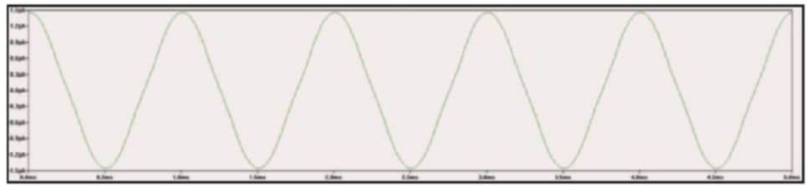


Figure 11: This control-grid current flows in the Class AB$_2$ circuit.

Figure 12: A true Class B amplifier would produce an output wave like this at a signal level near clipping.

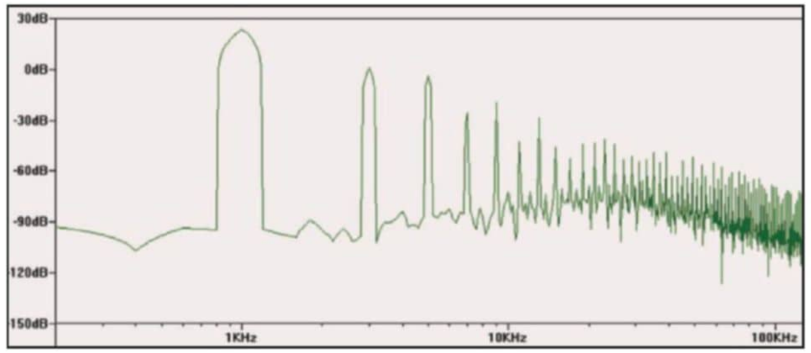


Figure 13: The FFT of the Class B amplifier clearly shows high levels of odd-order harmonics.

Class B Push-Pull

Although true Class B biasing is virtually never used in audio amplifiers, it is interesting to see what the output wave and its FFT would look like. Using a circuit like that of the Class AB$_1$ example, but with the output tubes control grids biased at cut-off, we get an output wave like that shown in **Figure 12**. Its FFT appears in **Figure 13**. It should be noted that with no negative feedback, the output wave of a Class B push-pull stage varies in waveform with the signal level, because the crossover notch represents a varying fraction of the instantaneous output voltage near the zero-crossing times. The example in Figure 12 represents the waveform just below the onset of clipping. The FFT of the Class B output wave shows very high levels of odd-order harmonics, compared with the waves from the amplifiers we've examined previously. Even-order harmonics are almost nonexistent. High-order harmonics are quite prominent.

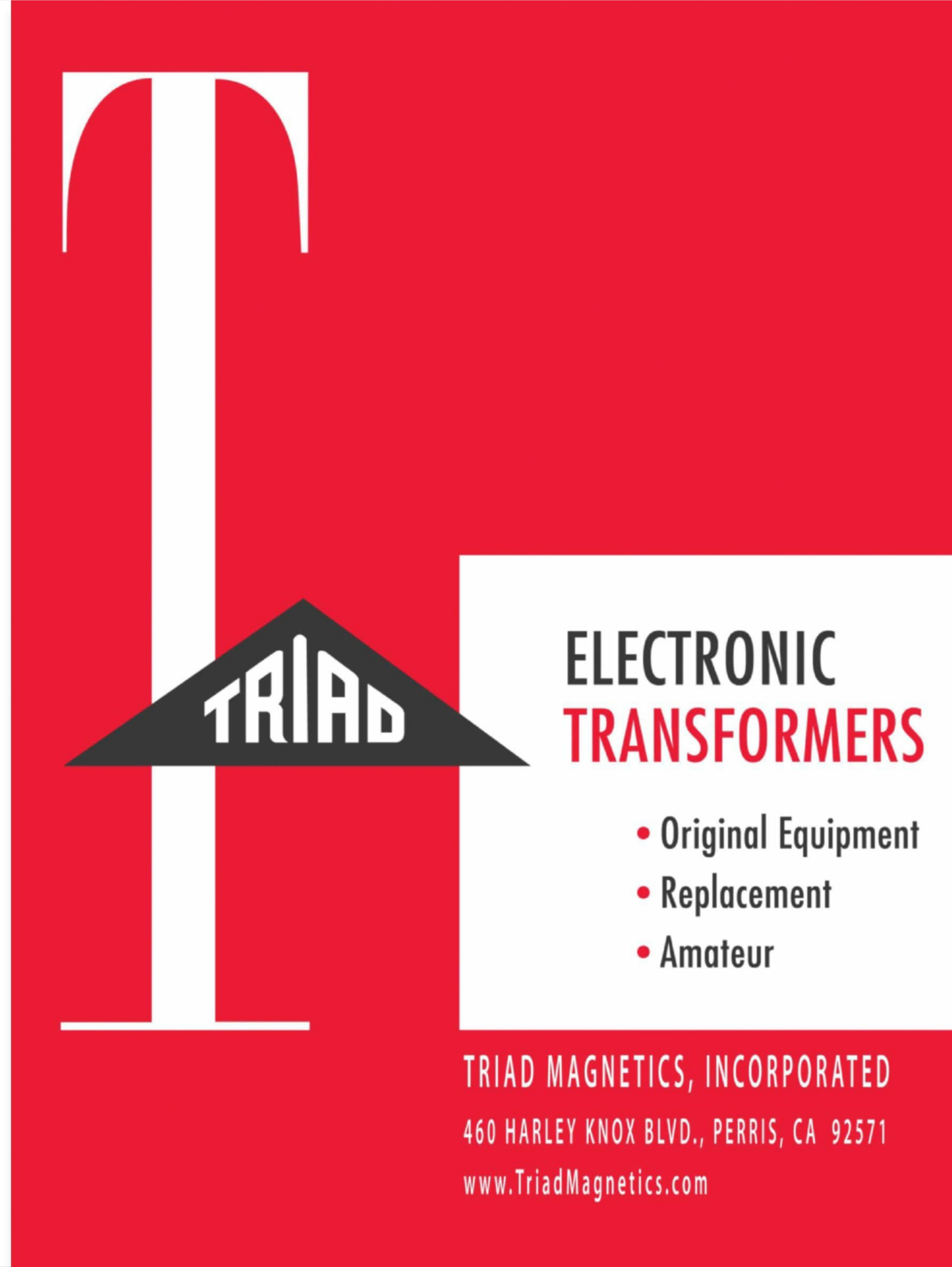
Summary

Class A audio amplifiers are most often found in single-ended designs. However, Class A biasing is also applicable to push-pull design, increasing the maximum output power from a pair of triodes or pentodes. With no negative feedback, Class A biasing provides the cleanest output of any push-pull design.

Class AB$_1$ designs have only slightly higher THD than Class A designs, but have operating efficiencies three-and-a-half times as high, permitting a sizeable increase in available output power from a given tube type. The THD is only about one percent higher than that of Class A push-pull, but that fact is misleading, because the Class AB$_1$ circuit has far higher high-order harmonics than does the Class AB$_1$ circuit, and these high-order harmonics are far more objectionable than the first-through-fifth-order harmonics that predominate in the output of a Class A circuit.

Class AB$_2$ amplifiers are slightly more efficient than Class AB$_1$ amps, but not enough to make a useful difference in output power. In fact, if you compare Table 2 and Table 3, you'll see that RCA's sample AB$_2$ circuit produces marginally less output power than the AB$_1$ circuit. But the plate current in the output tubes for Class AB$_2$ is a bit lower than for the Class AB$_1$ amp. This would lower the operating temperature of the power transformer in a Class AB$_2$ amp, which could be important in some applications.

Although true Class B amps are virtually never used in audio, we notice from the FFT that the circuit topology does in fact almost completely null out even-order harmonic distortion.



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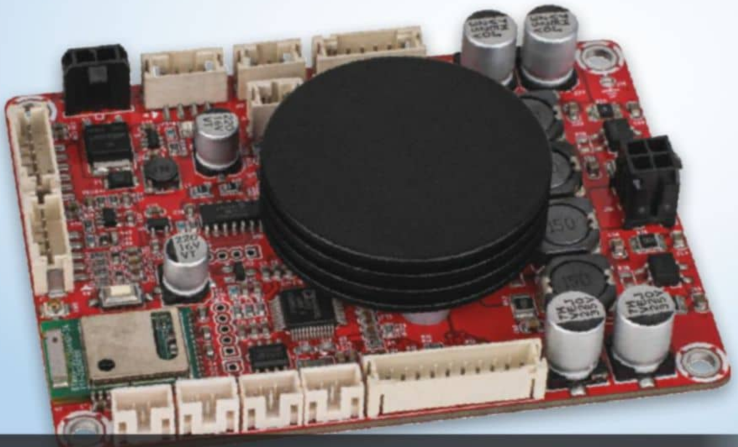

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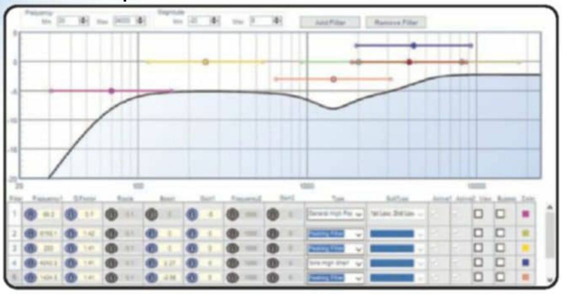

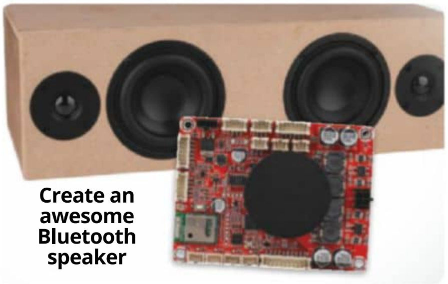

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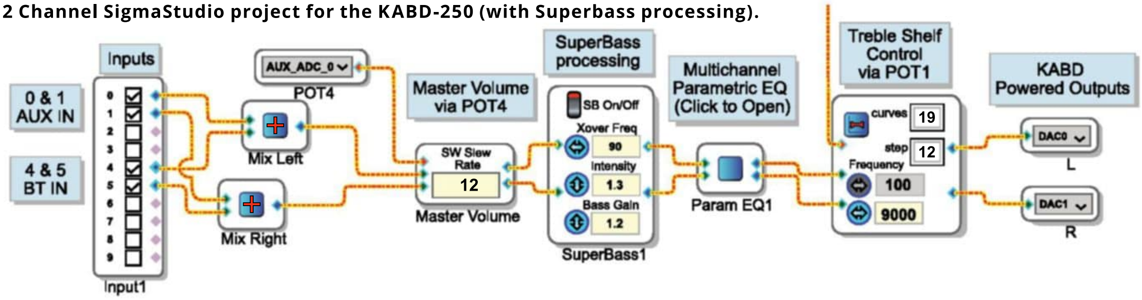

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